SIP Software for Avaya 1200 Series IP Deskphones-Administration

SIP3.2
NN43170-601, Standard 01.06
January 2012

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Contents

Chapter 1: New in this release

This document is new for SIP Release 3.2. This document contains administration information for the Avaya 1220 IP Deskphone and Avaya 1230 IP Deskphone.

Features

SIP Release 3.2 introduces User interface and preference enhancements to the IP Deskphone to allow for better usability and new functionality.

The following is a list of features that have been modified or added:

- Menu auto back-out time
- Inbox and outbox navigation modifications
- Customizable banner for login
- Phone Information Details screen
- Speed Dial List
- Busy Lamp Field

To provision User interface and preferences enhancements, see Device configuration commands on page 41. For more information about SIP Release 3.2 features, see the following sections:

Multiple Appearance Directory Number

SIP Release 3.2 introduces support for Multiple Appearance Directory Number (MADN) for Communication Server 1000. For more information, see Multiple Appearance Directory Number (Single Call Arrangement) on page 127.

Security

SIP Software Release 3.2 introduces many security features for the IP Deskphone including the presence of a security icon (padlock) that appears on the screen of the IP Deskphone to indicate that the media path of a call is encrypted.

Additional security features include the following:

- SIP over TLS
- Connection persistence
- SSH and Secure File Transfer
- SRTP/SRTCP and SDESC

- Last successful or unsuccessful logonLast successful or unsuccessful logon
- Enhanced administrative password security

To provision the Security feature, see Security on page 237.

Other changes

There are no other changes.

Revision history

January 2012	Standard 01.06. This document is up-issued to reflect changes in technical content for Trusted Root Certificate installation.
May 2011	Standard 01.05. This document is up-issued to reflect changes in technical content for the following: • AUTOLOGIN_ID_KEY parameters • reset codecs to default • modifying the SIP provisioning file
March 2011	Standard 01.04. This document is up-issued to reflect changes in technical content for SIP and UNIStim software versions and for changes to the AUTOCLEAR_NEWCALL feature configuration command.
October 2010	Standard 01.03. This document is up-issued to reflect changes in the configuration of TLS for SIP.
September 2010	Standard 01.02. This document is up-issued with minor revisions.
August 2010	Standard 01.01. This document is a new document and is issued to support SIP Release 3.2.

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Navigation

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Chapter 3: Introduction to this guide

Subject

This document describes how to install, configure, and provision the Avaya 1220 IP Deskphone and Avaya 1230 IP Deskphone for use on a SIP network. The Avaya 1220 IP Deskphone and Avaya 1230 IP Deskphone are collectively known as Avaya 1200 Series IP Deskphones. In this document, the Avaya 1220 IP Deskphone and Avaya 1230 IP Deskphone are referred to as IP Deskphones.

Intended audience

This administration guide is intended for system administrators of the Avaya 1220 IP Deskphone and Avaya 1230 IP Deskphone with a basic understanding of SIP. This guide is not intended for end users of the Avaya 1220 IP Deskphone and Avaya 1230 IP Deskphone. Many of the tasks outlined in the guide influence the function of the IP Phone on the network and require an understanding of telephony and Internet Protocol (IP) networking.

Acronyms

This guide uses the following acronyms:

Table 1: Acronyms used

AAA	Authentication, Authorization, and Accounting
ALG	Application Layer Gateway
BER	Bit Error Rate
CA	Certificate Authority
CN	Common Name
CRL	Certificate Revocation List
CTL	Certificate Trust List

DCP	Device Certificate Profile
DET	Distinguished Encoding Rules
DHCP	Dynamic Host Configuration Protocol
DN	Distinguished Name
DND	Do Not Disturb feature
DNS	Domain Name System
DRegex	Digit Regular Expression
DSCP	Differentiated Services Code Point
EAP	Extensible Authentication Protocol
ECR	Error Collection and Recovery
EJBCA	Enterprise Java Bean Certificate Authority
ERE	Extended Regular Expressions
FQDN	Fully Qualified Domain Name
FTP	File Transfer Protocol
GARP	Gratuitous Address Resolution Protocol
GUI	Graphical User Interface
HTTP	Hyper Text Transfer Protocol
HTTPS	Hyper Text Transfer Protocol over SSL
IAS	Internet Authentication Service
ICMP	Internet Control Message Protocol
IETF	Internet Engineering Task Force
ISDN	Integrated Services Digital Network
IM	Instant Message
IP	Internet Protocol
IPCM	Internet Protocol Client Manager
ITU-T	Telecommunications Standardization sector of the International Telecommunications Union
LAN	Local Area Network
LED	Light Emitting Diode
MAC	Media Access Control
MADN	Multiple Appearance Directory Number
MAS	Media Application Server
MD5	Message Digest v5

NAT	Network Address Translator
NetConfig	Configuration screens available after an IP Deskphone resets
NDU	Network Diagnostic Utility
OAM	Operation, Administration (and) Maintenance
PDT	Problem Determination Tool
PEAP	Protected Extensible Authentication Protocol
PEC	Product Engineering Code
PKCS#12	Public Key Cryptographic Standard #12
POE	Power Over Ethernet
POSIX	Portable Operating System Interface
PRACK	Provisional Acknowledgement
PSTN	Public Switched Telephone Network
PVQMon	Proactive Voice Quality Monitoring
QoS	Quality of Service
RADIUS	Remote Authentication Dial In User Service
RTCP	Real-time Control Protocol
RTCP XR	RTP Control Protocol Extended Reports
RTP	Real-time Transfer Protocol
SAN	Subject Alternate Name
SCA	Single Call Arrangement Shared Call Appearance
SCEP	Simple Certificate Enrollment Protocol
SDP	Session Description Protocol
SFS	Security File System
SIMPLE	SIP for Instant Messaging and Presence Leveraging Extensions
SIP	Session Initiation Protocol
SKS	Special Key Sequence
SMTP	Simple Mail Transfer Protocol
SOAP	Simple Object Access Protocol
STUN	Simple Traversal of UDP through NAT devices
TCP	Transport Control Protocol
TFTP	Trivial File Transport Protocol
TLS	Transport Level Security

TPS	Terminal Proxy Server
TTL	Time-to-live
UDP	User Datagram Protocol
UFTP	UNIStim File Transfer Protocol
UI	User Interface
UNIStim	Unified Network IP Stimulus Protocol
VoIP	Voice over IP
VLAN ID	Virtual Local Area Network Identification
VLAN IP	Virtual Local Area Network Internet Protocol
VQMon	Voice Quality Monitoring

Related publications

Other publications related to the SIP Software for Avaya 1200 Series IP Deskphones administration include the following:

- *Avaya 1220 IP Deskphone with SIP Software User Guide , NN43170-101*
- *Avaya 1230 IP Deskphone with SIP Software User Guide , NN43170-102*
- *Broadsoft Partner Configuration Guide Avaya SIP Software for 1120E/1140E IP Deskphones, NN43121-300*
- *Avaya 1200 Series Expansion Module (SIP Software) User Guide, NN43170-103*
- Avaya 1200 Series IP Deskphones product bulletins on http://support.avaya.com/css/appmanager/public/support.

Chapter 4: Overview

Introduction

This chapter describes the hardware and software features of the Avaya 1200 Series IP Deskphones and provides a brief overview of Session Initiation Protocol (SIP). In this document, Avaya 1200 Series IP Deskphones are referred to as IP Deskphones.

SIP overview

Session Initiation Protocol (SIP) is a signaling protocol used for establishing multimedia sessions in an Internet Protocol (IP) network.

SIP is a text-based protocol similar to HTTP and SMTP. With the introduction of SIP to IP Deskphones, telephony integrates easily with other Internet services. SIP allows the convergence of voice and multimedia.

Avaya 1200 Series IP Deskphones with SIP Software

The Avaya 1200 Series IP Deskphones connect to an IP network using an Ethernet connection. All voice and signaling information is converted into IP packets and sent across the network.

IP Deskphones can be ordered with UNIStim software installed or with SIP software installed. UNIStim software and SIP software use the same hardware, but the order number of an IP Deskphone with UNIStim software is different from the order number of an IP Deskphone with SIP software.

If you have an IP Deskphone with UNIStim software, you can convert the software to SIP software. To download the most recent version of SIP software, see Download the SIP Software to the provisioning server on page 32.

This guide explains how to:

- configure the provisioning server and the DHCP server. Note: The provisioning server is where the software and the configuration files for the IP Deskphones reside. This is not the IP Client Manager (IPCM) of the Call Server.
- convert an IP Deskphone with UNIStim software to an IP Deskphone with SIP software
- provision the Device Settings parameters on the IP Deskphones with SIP software

Important:

Converting the software on an IP Deskphone from UNIStim software to SIP software overwrites the UNIStim software. The IP Deskphone cannot operate in both modes simultaneously. A switch from UNIStim to SIP software or SIP to UNIStim software requires a software reload.

The following figure shows the main components of the Avaya 1230 IP Deskphone with SIP software.

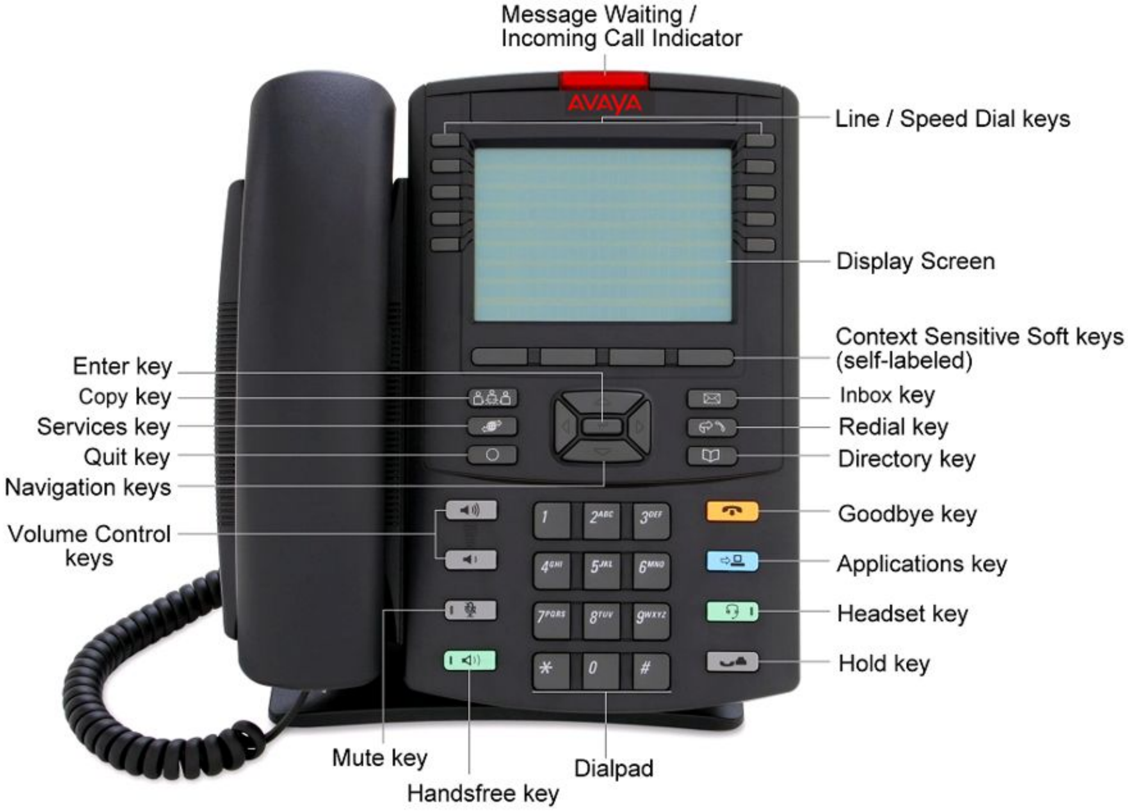


Figure 1: Avaya 1230 IP Deskphone with SIP Software

Related documentation

The Avaya 1200 Series IP Deskphones with SIP Softphone User Guide explains how to do the following:

- use the context-sensitive soft keys and Navigation key cluster
- enter text
- use the address book
- access and use the call inbox and call outbox
- configure and use instant messaging
- receive, identify, answer, redirect, decline, or ignore an incoming call
- operate hold, three-way calling, call transfer, and call park
- use other features such as speed dial, call forward, do not disturb, and setting up conference calls
- use the Multi line appearance/Bridged line appearance feature

For more information about using the IP Deskphones, see *Avaya 1220 IP Deskphone with SIP Software User Guide, NN43170-101*, *Avaya 1230 IP Deskphone with SIP Software User Guide , NN43170-102* .

The Avaya 1200 Series IP Deskphones Getting Started Card included in the box with the IP Deskphone explains how to do the following:

- connect the AC power adapter
- control the volume when answering a call
- make a call using the handset
- make a call with the headset or using handsfree
- use hold and mute
- set the contrast
- set the language

Installation overview

To install the IP Deskphone with SIP Software, three basic steps are required.

1. Configure the provisioning server and, optionally, the DHCP server. The function of the provisioning server is to provide configuration options to every IP Deskphone throughout the network. The DHCP server can be configured to provide basic

network-configuration data or a more comprehensive set of network-configuration data for the IP Deskphone with SIP Software.

2. Load SIP Software on the IP Deskphone.
3. Configure the initial network-configuration parameters on the IP Deskphone with SIP Software.

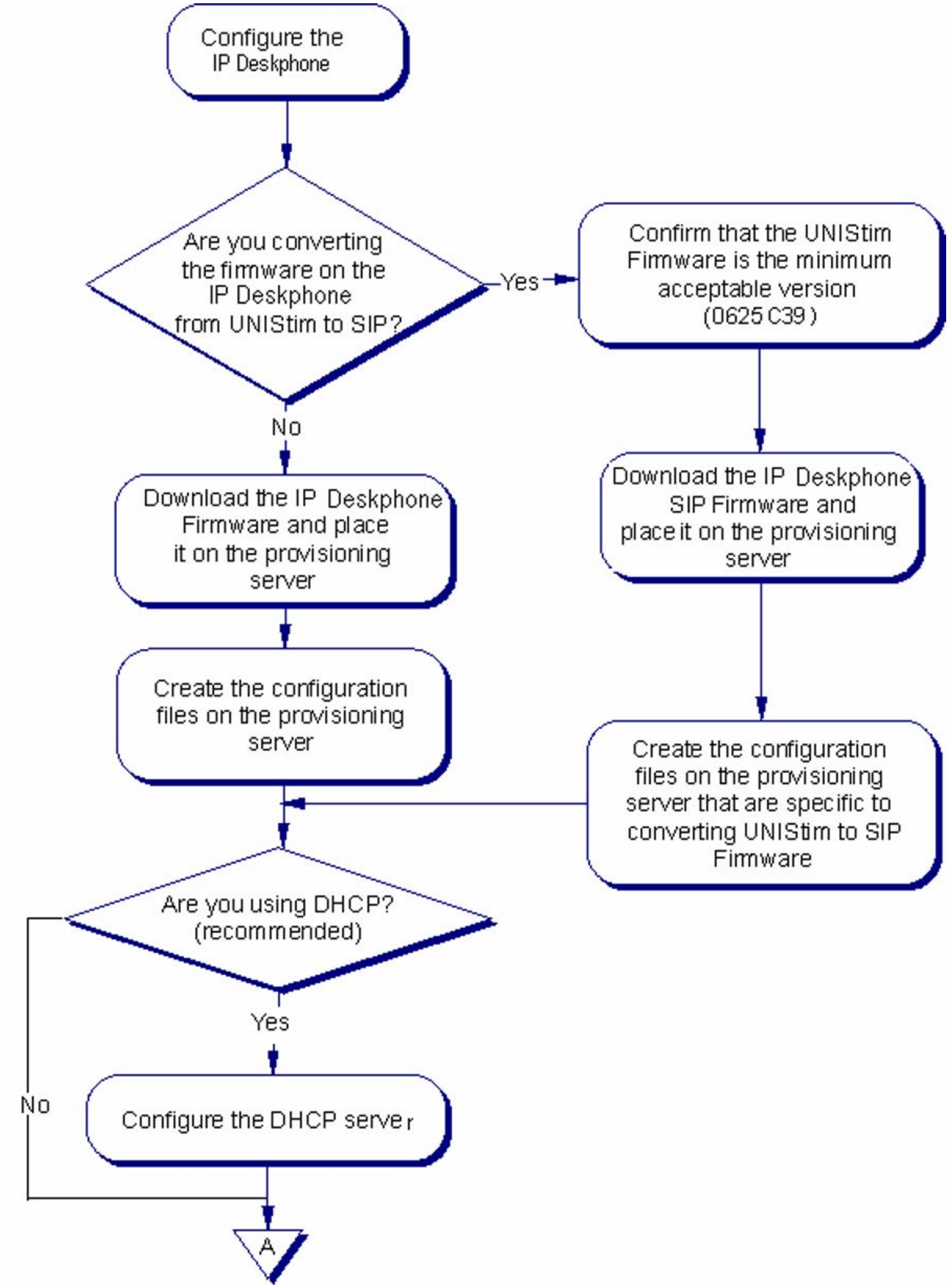

Configure the IP Deskphone
Are you converting the fimware on the IP Deskphone from UNIStim to SIP?
Yes
Confirm that the UNIStim Fimware is the minimum acceptable version (0625 C39)
No
Download the IP Deskphone Firmware and place it on the provisioning server
Download the IP Deskphone SIP Firmware and place it on the provisioning server
Create the configuration files on the provisioning server
Create the configuration files on the provisioning server that are specific to converting UNIStim to SIP Fimware
Are you using DHCP? (recommended)
Yes
No
Configure the DHCP server
A

Figure 2: Installation of Avaya 1230 IP Deskphone with SIP Software, page 1 of 2

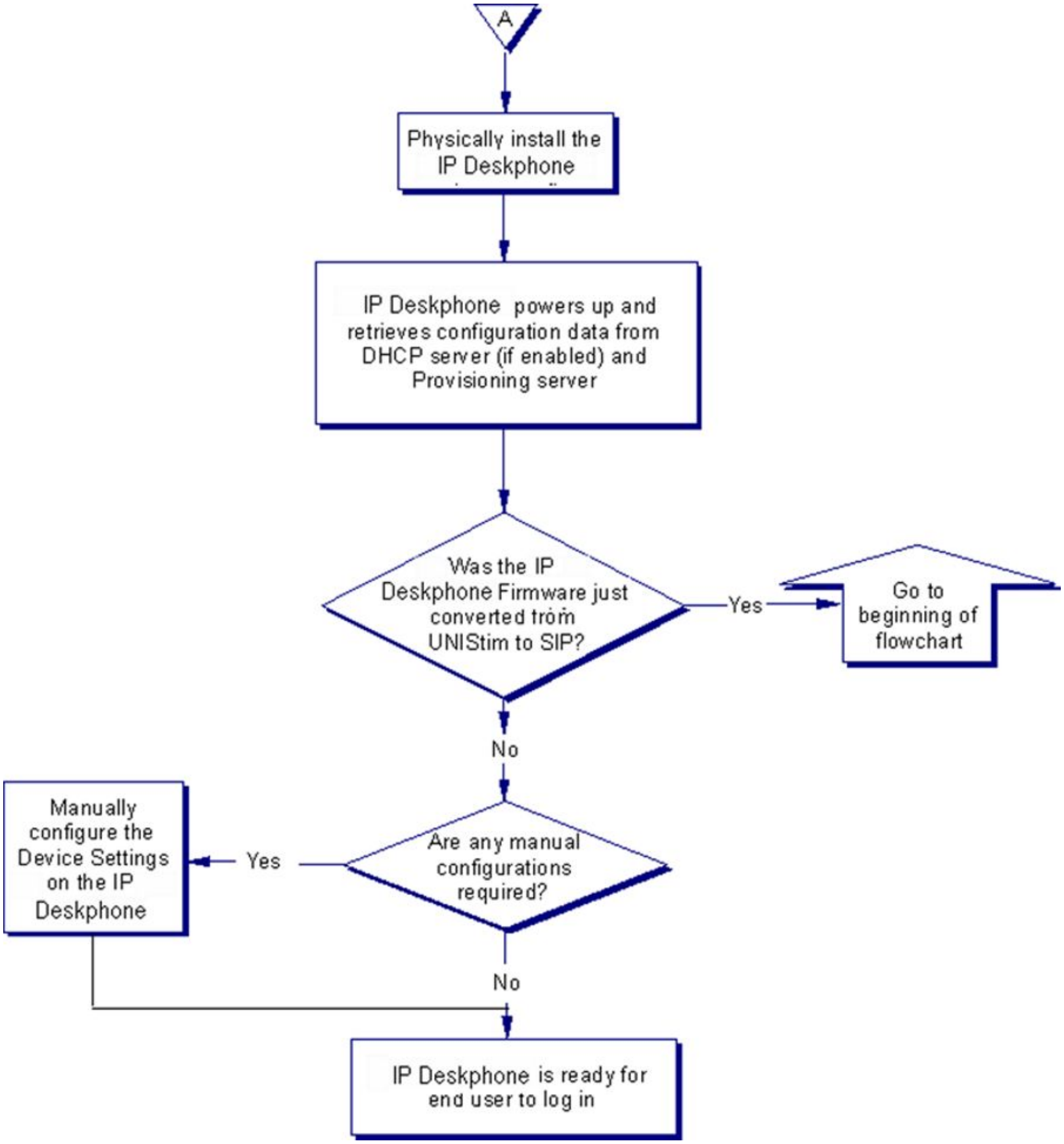

A
Physically install the IP Deskphone
IP Deskphone powers up and retrieves configuration data from DHCP server (if enabled) and Provisioning server
Was the IP Deskphone Firmware just converted from UNIStim to SIP?
Yes
Go to beginning of flowchart
No
Are any manual configurations required?
Yes
Manually configure the Device Settings on the IP Deskphone
No
IP Deskphone is ready for end user to log in

Figure 3: Installation of Avaya 1230 IP Deskphone with SIP Software, page 2 of 2

Chapter 5: Before installation

Introduction

This chapter features a checklist of tasks you must complete before you install SIP Software on the Avaya 1200 Series IP Deskphone.

Preinstallation

Complete the following checklist.

Preinstallation checklist

1. Read and become familiar with your IP Deskphone User Guide.
2. Ensure there is one IP Deskphone boxed package for each IP Deskphone being installed.
3. Ensure that the IP Deskphone box includes the following:

Table 2: IP Deskphone box contents

Item	Avaya 1220 IP Deskphone order number	Avaya 1230 IP Deskphone order number
IP Deskphone Graphite with icon keys without PS (SIP) (RoHS)		
IP Deskphone Graphite with English keys without PS (SIP) (RoHS)		
Handset, Charcoal	NTYS09AA70	
Handset cord, Charcoal	NTYS10AA70	
Footstand kit, Charcoal	NTYS11AA70	
Phone number label and lens kit	NTYS12AA	

Item	Avaya 1220 IP Deskphone order number	Avaya 1230 IP Deskphone order number
2.3 m (7 ft) CAT5 Ethernet cable	NTYS13AA	

The IP Deskphone can be powered either by Power Over Ethernet (POE) or through an external power supply. Order the external power supply separately.

Warning:

Do not use the AC power adapter, if you are connected to a Power over the Ethernet (PoE) connection. Only use the AC power adapter when you do not have a Power over the Ethernet connection.

Table 3: Avaya 1200 Series IP Deskphones parts list (order separately)

CPC code	PEC code	Product description
N0146475	NTYS17BAE6	IP Deskphone Global Power Supply (2000, 1100, 1200) (RoHS)
N0089603	NTYS14AAE6	Standard IEC Cable - North America (RoHS)
A0781922	NTTK15AA	Standard IEC Cable – Australia / NZ (Note: RoHS not required)
N0114986	NTTK16ABE6	Standard IEC Cable – Europe
N0109787	NTTK17ABE6	Standard IEC Cable – Switzerland
N0109881	NTTK18ABE6	Standard IEC Cable – UK
N010978	NTTK22ABE6	Standard IEC Cable – Denmark
A0814961	A0814961	Standard IEC Cable - Argentina (Note: RoHS not required)
N0118951	NTTK26AAE6	Standard IEC Cable - Japan

Caution:

The IP Deskphone must be plugged into a 10/100-BaseT Ethernet jack. Severe damage occurs if this IP Deskphone is plugged into an ISDN connection.

4. Ensure that the location meets the network requirements:
   - a DNS server and a DHCP server with DHCP relay agents installed, configured, and running. Using DHCP and DNS servers with CS 2000 network is recommended but not mandatory.
   - An Ethernet connection to a network with an appropriate SIP proxy server.

- One of the following file servers used as a Provisioning server:
    - TFTP server
    - FTP server
    - HTTP server

  Only a TFTP server can be used for an initial UNIStim-to-SIP Software conversion. An IP Deskphone with SIP Software can operate with a TFTP, FTP, or HTTP file server.

Chapter 6: Configure the provisioning server

Important:

If you have UNIStim software on your IP Deskphone, the software must be converted from UNIStim to SIP before you proceed with the following instructions. See the chapter Upgrade and convert the IP Deskphone software on page 89 for instructions on how to convert the software on an IP Deskphone from UNIStim to SIP.

If the IP Deskphone is installed with SIP Software, further SIP Software upgrades can be done with a TFTP, an FTP, or an HTTP server.

How provisioning works

Provisioning is performed without interaction with the Call Server. The Avaya 1200 Series IP Deskphone with SIP Software connects directly with the provisioning server in order to retrieve software files and configuration files. In this case, the provisioning server is not to be confused with the IP Client Manager on the Call Server. The methods of provisioning are as follows:

- Automatic provisioning at power-up: After the IP Deskphone powers up or is reset, it checks the provisioning server for the latest files.
- Provisioning through user interaction: The end user can manually check for updates by pressing the Services context-sensitive soft key and selecting Check for Updates.
- Automatic provisioning at a preconfigured time: The IP Deskphone with SIP Software checks for updates every 24 hours, at a time specified by a parameter in the device configuration file.

The following is the sequence of events when provisioning updates occur. The IP Deskphone with SIP Software:

1. connects to the provisioning server
2. retrieves the provisioning file (for example, 1230SIP.cfg) from the provisioning server
3. reads and acts upon the content of the provisioning file and decides whether any other file is needed, based on a set of rules. If files need to be downloaded to the IP Deskphone, a new file transfer session starts for each file to be downloaded. The provisioning file (for example, 1230SIP.cfg) can contain commands that prompt for confirmation before a file is downloaded.

Download the SIP Software to the provisioning server

To download the SIP Software, perform the following procedure.

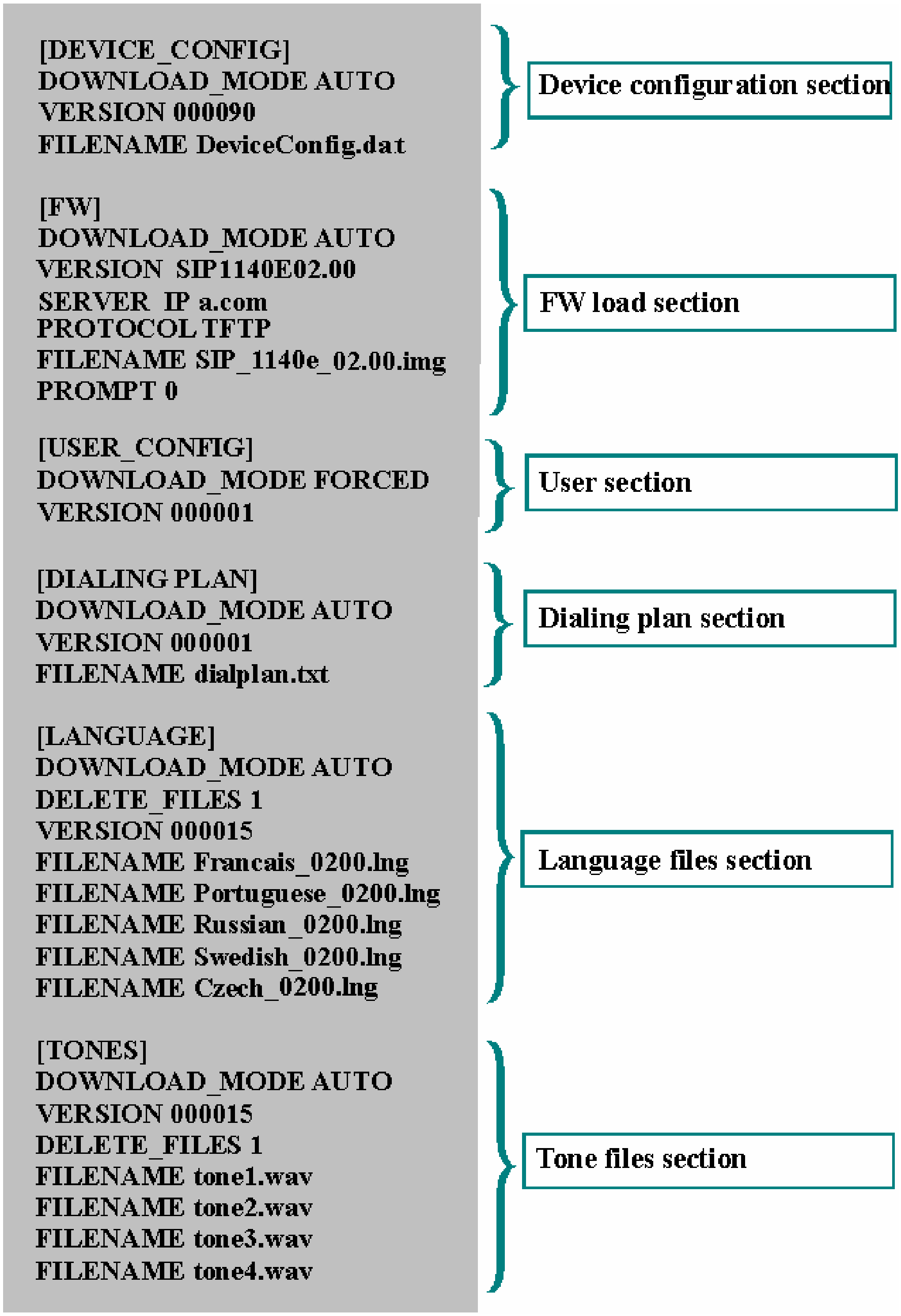
Downloading SIP Software for the IP Deskphone

1. Go to http://www.avaya.com/support.
2. Log on to the Avaya Web site with a valid Avaya User ID and Password.

   The **Support** page appears.
3. Enter the IP Deskphone type in the **Knowledge and Solution Engine** box.
4. Select **Software** in the **All types** scroll-down menu.
5. Press the gray arrow at the end of the **Knowledge and Solution Engine** box to obtain the **Search Results**.
6. From the **Search Results**, select and download the appropriate version of the SIP Software for the IP Deskphone; for example, **SIP IP Deskphone 1230 Release SIP12x004.01.03.00.bin**.
7. Place the selected software on the provisioning server.

Create the SIP provisioning file on the provisioning server

The provisioning file is downloaded from the provisioning server to the Avaya 1200 Series IP Deskphone every time the IP Deskphone checks for updates. The provisioning file is a clear text file that has the naming convention 12xxSIP.cfg. The following is an example of the IP Deskphone provisioning file:



```
[DEVICE_CONFIG]
DOWNLOAD_MODE AUTO
VERSION 000090
FILENAME DeviceConfig.dat

[FW]
DOWNLOAD_MODE AUTO
VERSION  SIP1140E02.00
SERVER  IP a.com
PROTOCOL TFTP
FILENAME SIP_1140e_02.00.img
PROMPT 0

[USER_CONFIG]
DOWNLOAD_MODE FORCED
VERSION 000001

[DIALING PLAN]
DOWNLOAD_MODE AUTO
VERSION 000001
FILENAME dialplan.txt

[LANGUAGE]
DOWNLOAD_MODE AUTO
DELETE_FILES 1
VERSION 000015
FILENAME Francais_0200.lng
FILENAME Portuguese_0200.lng
FILENAME Russian_0200.lng
FILENAME Swedish_0200.lng
FILENAME Czech_0200.lng

[TONES]
DOWNLOAD_MODE AUTO
VERSION 000015
DELETE_FILES 1
FILENAME tone1.wav
FILENAME tone2.wav
FILENAME tone3.wav
FILENAME tone4.wav
```

Figure 4: Sample provisioning file

Table 4: Provisioning file supported sections

[DEVICE_CONFIG]	Device configuration file
[FW]	Firmware image
[DIALING_PLAN]	Dialing plan
[LANGUAGE]	Downloadable language files (more than one can be specified in each section)
[IMAGES]	Downloadable images
[TONES]	Downloadable tones (.wav files)
[LICENSING]	License files
[POLICY]	License policy files
[DEV_CERT]	Device certification files
[USER_KEYS]	User keys files
[LOGIN_BANNER]	Login banner
[USER_CONFIG]	IP Deskphone-specific configuration files

Provisioning is performed using the commands in the 12xxSIP.cfg configuration file. The configuration file can have multiple sections.

Note:

The maximum length of a line item in the configuration file is 80 characters. If a line item with more than 80 characters is encountered when parsing the configuration file, the remaining portion of the file following that line item is ignored.

The '#' symbol is used to indicate a comment. Anything after a '#' symbol is a comment.

Each section in the configuration file defines rules for different file types. A section starts with a [SECTION NAME] to specify rules for each file type. For example: [FW].

A section is a mandatory field. Parsing of download rules for each file type starts with finding this key word. Currently, the following sections are supported by the IP Deskphone with SIP Software:

- [DEVICE_CONFIG] — used to configure various parameters in the IP Deskphone.
- [FW]—image files originate from Avaya only and are authenticated during software download. If the FW authentication fails, the IP Deskphone displays an error message and continues operation with the existing FW image.
- [DIALING_PLAN] — used for configuring dialing patterns and the format of originated URIs in the SIP message.
- [LANGUAGE] — simple text files containing all text prompts used by the IP Deskphone. Language files are used for the localization of the IP Deskphone without software upgrade. Each language file has a header that contains a software load version with which

this file is associated. Language files are signed by Avaya and are authenticated by the software for security reasons.

- [IMAGES] — section for the images files
- [TONES] — standard in the Telecommunications Standardization sector of the International Telecommunications Union (ITU-T). The IP Deskphone supports custom tone files. The tone files must be WAV files with the following specification: A-law or u-law (8.0 kHz, 8-bit, mono or 16.0 kHz, 16 bit mono). The WAV files can be created and downloaded to the IP Deskphone. These files are not authenticated by the IP Deskphone.
- [LICENSING] — section for the licensing files
- [SEC_POLICY] — section for downloading a file, which contains rules that define the security policy for the IP Deskphone. After the file downloads, the IP Deskphone verifies that the file is signed by a trusted entity before it accepts the values in the security policy file.
- [DEV_CERT] — section to enable an IP Deskphone to import the PKCS# 12 file.
- [CTL] — section to enable an IP Deskphone to download the Certificate Trust List.
- [USER_KEYS] — section to enable an IP Deskphone to download a customer root certificate.
- [LOGIN_BANNER]— section for the login banner files
- [USER_CONFIG] — section for IP Deskphone-specific configuration file

  IP Deskphone-specific configuration files support customizing the IP Deskphone on a per IP Deskphone/user level. Parameters in the device configuration file can be overwritten with a IP Deskphone-specific configuration file.

Mandatory keywords in the Provisioning file are:

- **VERSION [xxxxxx]**, where xxxxxx is a six- to ten-digit number representing the version of the file on the server. The version of the module is specified in this field. The command is used for version comparison in AUTO mode. VERSION is mandatory for all sections. In the FW section, the software version of the load located on the provisioning server must be entered in this field. For all other sections, VERSION is just a counter that can be incremented if it is necessary to download a new file version.

  ⚠ **Caution:**

  The version number is stored permanently on the IP Deskphone until a higher version number is downloaded. However, if the **Forced** option is in the 12xxSIP.cfg file, then the file is forced to download and the version number in the IP Deskphone is overwritten with the version number in the 12xxSIP.cfg file.

- **DOWNLOAD_MODE [AUTO | FORCED]** defines whether the version is checked. This command is optional. If this command is not present, AUTO mode is used as the default.
  - **AUTO** - This mode compares the version of the module from the VERSION field and the version of the module version stored in the FLASH memory of the IP Deskphone. The file download is initiated only if the version specified is higher than the current

version stored in the IP Deskphone. If the version is not applicable, as in the case of language files, the date of the file must be used for the decision.

Caution:

The version number stored in the FLASH is permanent until a higher number is downloaded from the Provisioning file or you select**Srvcs > System > Erase User Data** on the IP Deskphone.

- **FORCED** - This mode forces the software download process. FORCED can be used for software downgrade procedures.

Note:

In FORCED or AUTO DOWNLOAD_MODE, the version number is overwritten with each software download.

- **FILENAME [filename]** specifies the file name to be downloaded for this section. For the language and tone section, the use of multiple filenames is allowed.

Optional keywords in the Provisioning file are:

- **PROMPT [YES | NO]** is used to indicate if the IP Deskphone should prompt the user for an update before the operation is performed. This command is optional with the default configured as NO.
  - **YES** - enables the prompt
  - **NO** - disables the prompt
- **PROTOCOL [TFTP | FTP | HTTP]** defines the protocol used to download the file. The IP Deskphone with SIP Software supports TFTP, FTP and HTTP protocols for file download. This command is optional. If it is not present, the default protocol TFTP is used.

Important:

When using the TFTP protocol to transfer the software image, the average round trip time must be < 75 ms. The IP Deskphone times out and aborts the software download if the total download time is less than 10 minutes.

If the average round trip time is less than 75 ms, use the FTP or HTTP protocol to transfer the software image.

If using FTP or HTTP, then **SRV_USER_NAME** and **SRV_USER_PASS** are also key words. These commands specify the credentials used to log on to the file server for file download. If not present, the protocol default credentials are used (no credentials for TFTP and HTTP and anonymous with no password for FTP).

- **SERVER_IP [address]** allows the IP Deskphone to connect to the specified IP address or name of the server for which the file can be downloaded. If the IP address is not specified, the **SERVER_IP** that is used is the same SERVER_IP that is used to download the provisioning file.

- **DELETE_FILES [YES | NO]**, if present, erases the language and tone files stored in the IP Deskphone before new files are downloaded. Otherwise, new files with different names are added without erasing existing files. This command is optional. Note that there is a hard limit of 5 language files and 5 tone files that can be stored in the IP Deskphone. When the limits are exceeded, no new file can be accepted for download.
    - **YES** - erases the existing language and tone files
    - **NO** - does not erase existing language and tone files
- **SRV_USER_NAME [username]** - If the protocol is FTP or HTTP, this keyword specifies the user name to log on to the server.
- **SRV_USER_PASS [password]** - If the protocol is FTP or HTTP, this keyword specifies the password to log on to the server.

The downloading of these files is initiated when an IP Deskphone is powered on, when an automatic check for updates is invoked, or when you select **Srvcs, System, Erase User Data**. Any of these actions causes the IP Deskphone to contact the provisioning server and attempt to read the Provisioning file. A Soft Reset (**Srvcs > System > Reset Phone**) does not cause the IP Deskphone to retrieve the Provisioning file.

Setting the default language on the IP Deskphone to French

To configure the default language on a new IP Deskphone, or an IP Deskphone that has not been logged into by an end user, include the following in the [DEVICE_CONFIG] and [LANGUAGE] sections of the 12xxSIP.cfg configuration file.

[DEVICE_CONFIG] DOWNLOAD_MODE AUTO VERSION 000002 FILENAME DeviceConfig.dat

[LANGUAGE] DOWNLOAD_MODE AUTO VERSION 0000000001 FILENAME French_d24.lng

The DeviceConfig.cfg file should contain the following.

DeviceConfig.cfg DEF_LANG French_d24

On a new IP Deskphone, the language switches to French after downloading and processing the configuration files. The login menu displays in French. On a subsequent bootup, the login menu and all boot messages are in French.

For a new user login, the IP Deskphone creates a new user profile. All menus remain in French. When a new user is created, the default language used is obtained from the DeviceConfig setting and stored as a user preference, after which the user preference for language is always used.

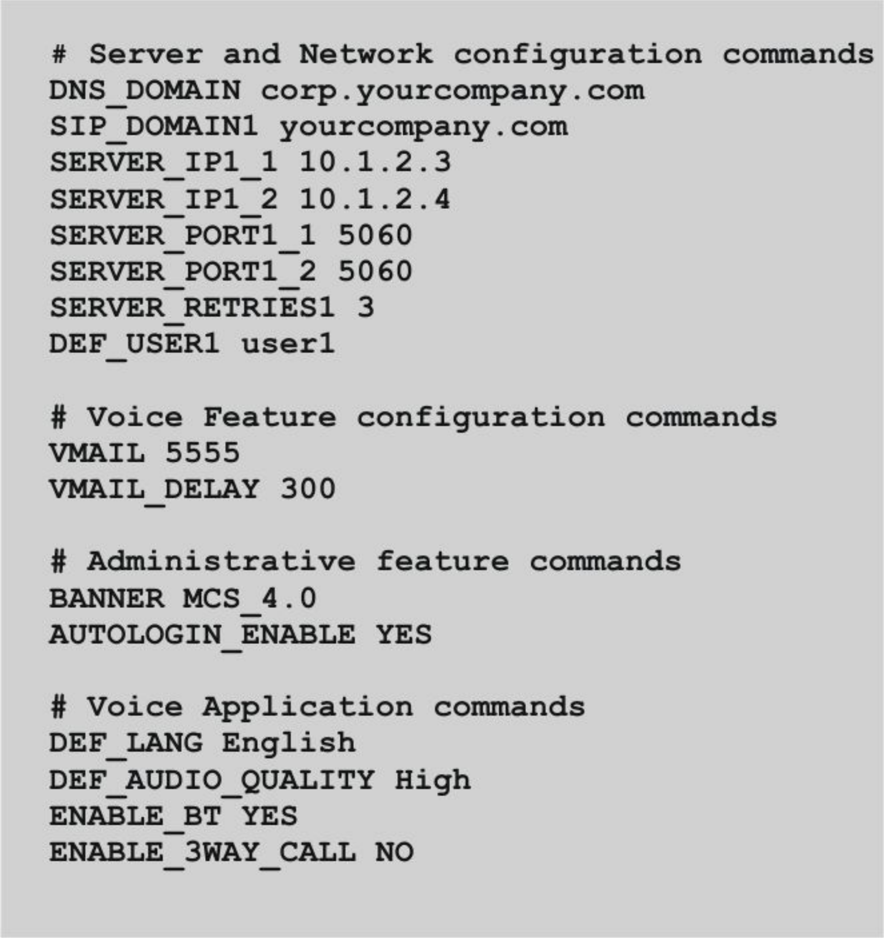
If a user has already logged in and either defaulted or chosen English as the user language preference, changing the configuration files does not affect the user's language display.

Create the device configuration file on the provisioning server

After the IP Deskphone downloads the provisioning file, the IP Deskphone reads the [DEVICE_CONFIG] section and is directed to download the device configuration file.

The device configuration file is a clear text file and the naming convention is defined by the administrator. See the FILENAME keyword in the [DEVICE_CONFIG] section of the SIP provisioning file.

The following is an example of a device configuration file.



```
# Server and Network configuration commands
DNS_DOMAIN corp.yourcompany.com
SIP_DOMAIN1 yourcompany.com
SERVER_IP1_1 10.1.2.3
SERVER_IP1_2 10.1.2.4
SERVER_PORT1_1 5060
SERVER_PORT1_2 5060
SERVER_RETRIES1 3
DEF_USER1 user1

# Voice Feature configuration commands
VMAIL 5555
VMAIL_DELAY 300

# Administrative feature commands
BANNER MCS_4.0
AUTOLOGIN_ENABLE YES

# Voice Application commands
DEF_LANG English
DEF_AUDIO_QUALITY High
ENABLE_BT YES
ENABLE_3WAY_CALL NO
```

Figure 5: Sample device configuration file

The next table provides a summary of the commands that can be used in the device configuration file. A description and the exact syntax of each command is given in Device configuration commands on page 41.

Table 5: Device configuration commands

Configuration command type	Configuration commands	
Server and network configuration commands	SIP_DOMAIN1 SIP_DOMAIN2 SIP_DOMAIN3 SIP_DOMAIN4 SIP_DOMAIN5 SERVER_IP1_1 SERVER_IP1_2 SERVER_IP2_1 SERVER_IP2_2 SERVER_IP3_1 SERVER_IP3_2 SERVER_IP4_1 SERVER_IP4_2 SERVER_IP5_1 SERVER_IP5_2 SERVER_PORT1_1 SERVER_PORT1_2 SERVER_PORT2_1 SERVER_PORT2_2 SERVER_PORT3_1	SERVER_PORT3_2 SERVER_PORT4_1 SERVER_PORT4_2 SERVER_PORT5_1 SERVER_PORT5_2 SERVER_RETRIES1 SERVER_RETRIES2 SERVER_RETRIES3 SERVER_RETRIES4 SERVER_RETRIES5 DNS_DOMAIN DEF_USERS DEF_USER1 DEF_USER2 DEF_USER3 DEF_USER4 DEF_USER5 UPDATE_USERS SIP_PING
Feature configuration commands	VMAIL VMAIL_DELAY AUTOLOGIN_ENABLE AUTOLOGIN_AUTHID_KEYxx PROMPT_AUTHNAME_ENABLE AUTO_UPDATE AUTO_UPDATE_TIME AUTO_UPDATE_TIME_RANGE TRANSFER_TYPE REDIRECT_TYPE ENABLE_PRACK SELECT_LAST_INCOMING	MAX_LOGINS MAX_INBOX_ENTRIES MAX_OUTBOX_ENTRIES MAX_REJECTREASONS MAX_CALLSUBJECT MAX_PRESENCENOTE DEF_LANG MAX_IM_ENTRIES MAX_ADDR_BOOK_ENTRIES ADDR_BOOK_MODE DEF_AUDIO_QUALITY
Feature configuration commands (continued)	PROXY_CHECKING ENABLE_BT AUTH_METHOD BANNER FORCE_BANNER DST_ENABLED TIMEZONE_OFFSET	HOLD_TYPE ENABLE_3WAY_CALL DISABLE_PRIVACY_UI DISABLE_OCT_ENDDIAL FORCE_OCT_ENDDIAL SNTP_ENABLE SNTP_SERVER

Configuration command type	Configuration commands	
	FORCE_TIME_ZONE IM_MODE IM_NOTIFY DEF_DISPLAY_IM CALL_WAITING DISTINCTIVE_RINGING USE_RPORT TOVM_SOFTKEY_ENABLE TOVM_VOICEMAIL_ALIAS TOVM_VOICEMAIL_PARAM MAX_RING_TIME ENABLE_UPDATE E911_USERNAME E911_PASSWORD	MADN_TIMER MADN_DIALOG DEFAULT_CFWD_NOTIFY FORCE_CFWD_NOTIFY RTP_MIN_PORT RTP_MAX_PORT SCA_HOLD_BEHAVIOR SCA_APPEARANCES SCA_BROADWORKS EXP_MODULE_ENABLE FORCE_REBOOT PROMPT_ON_LOCATION_OTHER E911_PROXY E911_TXLOC
Feature configuration commands (continued)	MENU_AUTO_BACKOUT AUTOCLEAR_NEWCALL_MSG LOGIN_BANNER_ENABLE SECURE_UI_ENABLE	BG_IMAGE_ENABLE BG_IMG_SELECT_ENABLE USE_BG_IMAGE SPEEDLIST_KEY_INDEX SPEEDLIST_LABEL BLF_ENABLE BLF_RESOURCE_LIST_URI FM_SOUNDS_ENABLE FM_IMAGES_ENABLE FM_CERTS_ENABLE FM_CONFIG_ENABLE FM_LOGS_ENABLE
Feature configuration commands (continued)	HOTLINE_ENABLE HOTLINE_URL SESSION_TIMER_ENABLE SESSION_TIMER_DEFAULT_SE SESSION_TIMER_MIN_SE	SET_REQ_REFRESHER SET_RESP_REFRESHER ENABLE_INTERWORKING MAX_ALLOWEDADDRESSES PORT_MIRROR_ENABLE MEMCHECK_PERIOD DOS_PACKET_RATE DOS_MAX_LIMIT DOS_LOCK_TIME LOGSIP_ENABLE
Feature configuration commands (continued)	CUST_CERT_ACCEPT CERT_ADMIN_UI_ENABLE SEC_POLICY_ACCEPT SECURITY_LOG_UI_ENABLE KEY_SIZE KEY_ALGORITHM TLS_CIPHER SIGN_SIP_CONFIG_FILES FP_PRESENTED	SUBJ_ALT_NAME_CHECK_ENABLE SECURITY_POLICY_PARAM_CHANGE CERT_EXPIRE AUTO_PRV_ACCEPT DWNLD_CFG_ACCEPT AUTO_PRV_SIGNING DWNLD_CFG_SIGNING

Configuration command type	Configuration commands	
	FP_ENTERED	FTP_PASSWORD
QoS and ToS commands	DSCP_CONTROL 802.1P_CONTROL DSCP_MEDIA	802.1P_MEDIA DSCP_DATA 802.1P_DATA
Tone configuration commands	DIAL_TONE RINGING_TONE BUSY_TONE	FASTBUSY_TONE CONGESTION_TONE
NAT configuration commands	NAT_SIGNALLING NAT_MEDIA NAT_TTL	STUN_SERVER_IP1 STUN_SERVER_IP2 STUN_SERVER_PORT1 STUN_SERVER_PORT2
Voice Quality Monitoring (VQMon) configuration commands	VQMON_PUBLISH VQMON_PUBLISH_IP LISTENING_R_ENABLE LISTENING_R_WARN LISTENING_R_EXCE PACKET_LOSS_ENABLE PACKET_LOSS_WARN	PACKET_LOSS_EXCE JITTER_ENABLE JITTER_WARN JITTER_EXCE DELAY_ENABLE DELAY_WARN DELAY_EXCE SESSION_RPT_EN SESSION_RPT_INT
System commands	ADMIN_PASSWORD	
Audio Codecs	G729_ENABLE_ANNEXB G723_ENABLE_ANNEXA	
Deskphone Recovery command	RECOVERY_LEVEL	

Device configuration commands

Caution:

The syntax of the device configuration file is case sensitive. Verify that the commands entered follow the case defined in this document.

Important:

Parameters in the device configuration file with empty values are not allowed and cause write failure.

Server and network configuration commands

- SIP_DOMAIN[x] [domain_name] preconfigures the proxy domain name for all servers. The same configuration can be done through the domain configuration menu on the IP Deskphone.

  - x - the number of the SIP domain number from 1 to 5.

  - domain_name - the proxy domain name for all servers.

  Note:
  SIP_DOMAIN[x] is provisioned after user logout.

- SERVER_IP[x]_[y]_ip_address] configures the primary and secondary IP address for each domain, two proxies for each domain.

  - x - the domain number from 1 to 5.

  - y - the corresponding primary and secondary IP addresses. y=1 indicates the primary address and y=2 indicates the secondary address.

  - ip_address - the IP address of the SIP proxy server.

- SERVER_PORT[x]_[y] [port_number] configures the signaling ports for each proxy.

  - x - the domain number.

  - y - the corresponding primary and secondary IP addresses. y=1 indicates the primary address and y=2 indicates the secondary address.

  - port_number - the SIP proxy signaling port (default is 5060).

- SERVER_RETRIES[x] [number_of_retries] confirms the number of retries for each domain. The default number of retries is 3.

  - x - the domain number from 1 to 5.

  - number_of_retries - the number of retry attempts to connect to the proxy server.

- DNS_DOMAIN [domain] is the DNS domain of the IP Deskphone.

- DEF_USERS[x] [user_name] allows you to enter the default user name for all domains. When the device configuration file gets downloaded, the default user name is used when logging in.

  - x - the domain number from 1 to 5.

  - user_name - the default user name.

- UPDATE_USERS [YES | NO] affects the default user names stored in the IP Deskphone. If this flag is configured as YES, the default user names are overwritten each time a new device configuration file is downloaded.
  - YES - the default user names are overwritten each time a new device configuration file is downloaded.
  - NO - the default user names are not overwritten each time a new device configuration file is downloaded.
- SIP_PING [YES | NO] The SIP_PING configuration value is used to maintain server heartbeat detection and to keep a firewall pinhole open.

  When used for server heartbeat detection, the IP Deskphone periodically pings the SIP Proxy and awaits a response. When three attempts to ping the SIP Proxy fail, the IP Deskphone begins a failover process and attempts to connect to the next configured SIP Proxy IP in the same domain.

  When a NAT TRAVERSAL method is selected, the SIP_PING configuration value also helps keep a firewall pinhole open.

  Important:
  Decide carefully whether SIP_PING usage is appropriate for your environment. Even when SIP_PING is not used for NAT TRAVERSAL, it is highly likely that you must keep SIP_PING enabled for server heartbeat detection.

  If the IP Deskphone is behind a firewall, it is very likely that you must keep SIP_PING enabled, unless an alternate method of keeping the firewall pinhole open is used.

  The default value is YES if not specified in the device configuration file. If SIP_PING is changed in the Device configuration file, the IP Deskphone must be rebooted for the change to take effect.
  - YES - enables pinging
  - NO - disables pinging

Feature configuration commands

- **TOVM_SOFTKEY_ENABLE [YES | NO]**
  - **YES** - enables the toVM soft key on the IP Deskphone.
  - **NO** - disables the toVM soft key on the IP Deskphone.
- **TOVM_VOICEMAIL_ALIAS <string>** — customizes the user ID of the SIP URI of the voice mail system.

- **TOVM_VOICEMAIL_PARAM<string>** — customizes the parameter name of the SIP URI of the voice mail system.
- **SCA_APPEARANCES** — configures the maximum number of appearances used for outgoing calls by the Shared Call Appearance (SCA) group. The valid range for this parameter is 2 to 24. The default value is 12.
- **SCA_HOLD_BEHAVIOR [PRIVATE | PUBLIC]** — configures the default behavior of the Hold button when user-determined behavior does not exist. When a user creates a new profile, the default behavior is taken from this setting. After the creation of a new profile, this configuration setting is not used. The default option is PUBLIC.
- **RTP_MIN_PORT**

  The minimum RTP port value is an integer between 1024 and 65535, exclusive of the restricted SIP ports between 5059 and 5080. The default value is 50000.
- **RTP_MAX_PORT**

  The maximum RTP port value is an integer between 1024 and 65535, exclusive of the restricted SIP ports between 5059 and 5080. The default value is 50100.

  Note:

  The RTP port configuration parameters must satisfy the constraints that (RTP_MAX_PORT - RTP_MIN_PORT) is greater than or equal to 10 and less than 1000.

  Note:

  If there is a provisioning error, RTP_MIN_PORT is reset to the default value of 50000 and RTP_MAX_PORT is reset to the default value of 50100. An error message is logged. The SystemConfig file stores 50000 and 50100, rather than the erroneous configuration values, to indicate that the configuration attempt has been rejected.
- **CALL_WAITING [SPEAKER | STREAM]**
  - **SPEAKER** - the call waiting tone is played on the IP Deskphone speaker. This is the default option.
  - **STREAM** - the call waiting tone is injected into the stream played on the transducer in use for the active call
- **DISTINCTIVE_RINGING [YES | NO]**
  - **YES** - turns on the distinctive ringing feature. This is the default option.
  - **NO** - turns off the distinctive ringing feature.
- **USE_RPORT [YES | NO]**
  - **YES** - allows the IP Deskphone to work from behind and/or in front of a symmetrical NAT with servers and/or clients that support RFC3581.
  - **NO** - disables implementation of support for RFC3581. This is the default option.

Note:

To provision USE_RPORT, the IP Deskphone must be rebooted after the device configuration file is updated. To force a hard reboot after the device configuration file is updated, configure FORCE_REBOOT YES.

- **EXP_MODULE_ENABLE [YES | NO]**
  - **YES** - the IP Deskphone detects and enables an expansion module.
  - **NO** - the IP Deskphone does not detect an expansion module. This is the default option.
- **MAX_RING_TIME [x]** — an integer between 30 and 600 that configures the number of seconds for incoming calls to ring before ignoring them. The default value is 120.
- **ENABLE_UPDATE [YES | NO]**
  - **YES** - enables UPDATE message support and adds “UPDATE” to ALLOW header. This is the default option.
  - **NO** - disables UPDATE message support.

Note:

ENABLE_UPDATE is provisioned after user logoff.

- **FORCE_REBOOT [YES | NO]**
  - **YES** - forces hard reboot after device configuration update.
  - **NO** - does not force hard reboot after device configuration update. This is the default option.

Note:

In order for FORCE_REBOOT to reboot the IP Deskphone, the VERSION of the device configuration file must be incremented, even if DOWNLOAD_MODE is configured as FORCED.

- **PROMPT_ON_LOCATION_OTHER [YES | NO]**
  - **YES** - prompt the user to select new location if location “other” was previously selected.
  - **NO** - do not prompt the user to select new location if location “other” was previously selected. This is the default option.
- **VMAIL [vmail_number]** — is the voice mail address, which can be the URI or the DN number of the voice mail server. This command takes a string as a parameter. This is the default link for a new user profile only. Individual users can customize the link through **Prefs > Message Options > Voice Mail Settings**. This command has no effect on the user profiles after it is created.

  vmail_number - the number or URI of the voicemail server.

- **VMAIL_DELAY[x]** is a delay, configured in milliseconds, between when the voice mail server answers the call and the start of dialing the voice mail user ID. The default value is 1000ms.

  x - the delay in milliseconds

- **AUTOLOGIN_ENABLE [YES | NO | USE_AUTOLOGIN_ID] or [1 | 0 | 2]** — controls whether the IP Deskphone attempts to automatically log on to the proxy server.

  - **YES** - turns on the auto login feature.
  - **NO** - turns off the auto login feature.
  - **USE_AUTOLOGIN_ID** - enables the Auto Login ID feature using the userid specified in AUTOLOGIN_ID_KEY01 and the password specified in AUTOLOGIN_PASSWD_KEY01 to register and authenticate. Both userid and password must be specified.

    The AUTOLOGIN_ID_KEY01 and AUTOLOGIN_PASSWD_KEY01 parameters are defined in the IP Deskphone-specific configuration file.

    Note:
    When using this setting, the user is prevented from logging off the IP Deskphone.

  or

  - **1** - turns on the Autologin feature.
  - **0** - turns off the Autologin feature.
  - **2** - enables the Autologin ID feature using the User ID specified in AUTOLOGIN_ID_KEY01 and the password specified in AUTOLOGIN_PASSWD_KEY01 to register and authenticate. Both userid and password must be specified.

    The AUTOLOGIN_ID_KEY01 and AUTOLOGIN_PASSWD_KEY01 parameters are defined in the IP Deskphone-specific configuration file.

    Note:
    When using this setting, the user is prevented from logging off the IP Deskphone.

  Note:
  If Autologin ID is enabled in the IP Deskphone-specific configuration file, it is recommended that AUTOLOGIN_ENABLE be configured as either Yes/No or 1/0 in the device configuration file. This recommendation facilitates migrating an IP Deskphone that uses the IP Deskphone-specific configuration file to not using the IP Deskphone-specific configuration file. The migration to just using the device configuration file can be done by deleting the IP Deskphone-specific configuration file. If the device configuration file does not have the matching parameters in the IP Deskphone-specific configuration file, the IP Deskphone continues to use the previously assigned settings after the IP Deskphone-specific configuration file is

deleted. This recommendation applies to other parameters in the IP Deskphone-specific configuration file.

- **AUTOLOGIN_AUTHID_KEYxx** — is used for auto login when the AUTOLOGIN_ENABLE method is configured to USE_AUTOLOGIN_ID.

  If the config file does not contain AUTOLOGIN_AUTHID_KEYxx, the client uses the value from AUTOLOGIN_ID_KEYxx.

- **PROMPT_AUTHNAME_ENABLE** — is used to determine if the authentication ID screen is presented to the user. The default value is NO.

  YES - after the user login name is entered, the authentication ID screen appears.

  NO - after the user login name is entered, the password screen appears.

- **AUTO_UPDATE [YES | NO]** — is a command to enable or disable the automatic updating of the IP Deskphone configuration files from the provisioning server. Enabling this command causes the IP Deskphone to check for updates once every day. The default is disabled.

  - **YES** - turns on the AUTO_UPDATE feature.
  - **NO** - turns off the AUTO_UPDATE feature.

Note:

If the IP Deskphone encounters any Major or Critical error in memory during the Auto update process, the IP Deskphone reboots based on the recovery level set.

- **AUTO_UPDATE_TIME [x]** — is the actual time in seconds, starting from midnight, before an automatic update occurs. Each IP Deskphone adds random numbers to the time specified by this command so every IP Deskphone does not try to access the provisioning server at the same time. By default the automatic update feature is disabled (see AUTO_UPDATE_RANGE).

  x - the time after midnight that the automatic update occurs.

- **AUTO_UPDATE_TIME_RANGE [x]** — is the range in hours, from the AUTO_UPDATE_TIME where an IP Deskphone checks for updates from the server. The default range is 1 hour.

  x - the range in hours when the IP Deskphone checks for updates from the server. The range can be from 1 to 6 hours.

- **TRANSFER_TYPE [MCS | STANDARD]** — is used to configure the IP Deskphone to activate Avaya conference server-assisted attended transfers, instead of the industry standard method of attended transfers. The default setting is Standard.

  - **MCS** - the typical attended transfer used by Avaya proxies. MCS uses a conference server to do the attended transfer.
  - **STANDARD** - the standard method of a transfer. This method does not involve a conference server.

- **REDIRECT_TYPE [MCS | RFC3261]** — is a command used to select different protocols for IP Deskphone redirection. The default setting is MCS.
    - **MCS** - when the IP Deskphone receives either 301 (moved permanently) or 302 (moved temporarily) during registration, it is assumed the IP Deskphone is moved to a new system (proxy+registrar) and all subsequent messages are sent to the new address.
    - **RFC3261** - the IP Deskphone assumes that, if during registration, a 301 (moved permanently) is received, the message contains a new registrar address. The IP Deskphone tries to register to the registrar using the existing proxy.
- **ENABLE_PRACK [YES | NO]** — PRACK is utilized to make some SIP messages reliable and requires that an ACK be sent with many SIP messages. ENABLE_PRACK is often utilized to verify that early media is being received. See RFC3262 for details.

  Note:
  ENABLE_PRACK must be configured as NO when connected to the MCS 5100 Release 3.5 system.

  Note:
  ENABLE_PRACK is provisioned after user logoff.

    - **NO** - disables PRACK and is the default value.
    - **YES** - enables PRACK.
- **ENABLE_INTERWORKING [YES | NO]** — this command is used to enable the interworking feature to pre-authorize users or groups of users to access automatic call answer. The configuration values are YES and NO. The default value is NO.
    - **NO** - the interworking feature is disabled.
    - **YES** - the interworking feature is enabled.
- **ENABLE_ALLOWEDADDRESS** — limits the size of the list of user and domain addresses stored for auto-answered authorization. The default value is 100.
- **PROXY_CHECKING [YES | NO]** — enables and disables extra security checking when incoming requests are sent to the IP Deskphone. The IP Deskphone with SIP Software always sends requests through an outgoing proxy. However, it is possible, through this configuration, to be able to accept an incoming request directly or through an incoming proxy.
    - **YES** - means that the request must come directly from the proxy server. YES is the default to enable proxy checking.
    - **NO** - means the request can be sent directly to the IP Deskphone. (NO is only suitable in a few situations).

- **AudioCodec <n><codec id><description>** — is a command that specifies the codecs that are available for the user to select. You can configure up to 15 codecs.
  - **n** - means the codec number. The value is 1 to 15.
  - **codec ID** - means the codec identifiers are as follows:
    - PCMA
    - PCMU
    - G729
    - G723
  - **text description** - a text description of the codec. For more information about audio codec configuration, see Audio codecs on page 191
- **DEF_AUDIO_QUALITY [Low | Medium | High]** — is a command used to configure the default audio quality used for each new call. Audio quality can be changed when the call is active. If this command is not present in the configuration file, the IP Deskphone uses High quality as its default value. The possible parameters for this command are High, Medium, and Low. If any other parameter is entered or these commands are misspelled, the IP Deskphone uses High as the default setting.

  The following codecs are used for each selection:

  - **Low** - G729 ptime 30.
  - **Medium** - G711 ptime 30.
  - **High** - G711 ptime 20.
- **AUTH_METHOD [AUTH | AUTH_INT]** — is used to configure the SIP authentication method.
  - **AUTH** - only authenticates (username/password)
  - **AUTH_INT** - authentication plus integrity checking (an MD5 hash of the entity is also computed and checked).
- **BANNER [banner_text]** — preconfigures the banner on the IP Deskphone. Use a text string to configure the banner. For example, BANNER ABC Company configures the banner to ABC Company. The text string can have a maximum of 24 characters.

  banner_text - an ASCII string displayed on the screen of the IP Deskphone with SIP Software.
- **FORCE_BANNER [YES | NO]** — is configured by the system administrator through the configuration file. If FORCE_BANNER is configured as YES, the banner from the configuration file is reloaded each time the IP Deskphone powers up, even if the user changes the banner manually.
  - **YES** - causes the banner configured by the administrator to override any banner configured by the user.
  - **NO** - allows the user to configure the banner.

- **DST_ENABLED [YES | NO]** — enables and disables the Daylight Savings Time (DST) mechanism. The time received from the server is GMT and is converted to the proper timezone by the IP Deskphone. If the Daylight Savings Time feature is enabled, the IP Deskphone automatically calculates the DST time at the appropriate date and converts the time to and from DST. The calculations used are based on the new rules applicable to DST in 2007. The IP Deskphone is programmed to use the North American DST scheme.

  - **YES** - enables Daylight Savings Time.

  - **NO** - disables Daylight Savings Time.

- **TIMEZONE_OFFSET [x]** — is used to configure the current time zone offset from GMT in seconds. TIMEZONE_OFFSET takes a number as a parameter. For example, TIMEZONE_OFFSET -25200 configures the time zone offset to MST, which is GMT-7 (-7*3600 = -25200 seconds).

Table 6: Time zone offset

Location	Time zone offset (seconds)
(GMT-10:00) Hawaii	-36000
(GMT-09:00) Alaska	-32400
(GMT-08:00) Pacific time (US and Canada)	-28800
(GMT-07:00) Mountain time (US and Canada)	-25200
(GMT-06:00) Central time (US and Canada)	-21600
(GMT-05:00) Eastern time (US and Canada)	-18000
(GMT-04:00) Atlantic time (US and Canada)	-14400
(GMT-03:00) Brasilia, Buenos Aires	-10800
(GMT+00:00) Greenwich, Dublin, Lisbon, London	0
(GMT+01:00) Amsterdam, Berlin, Rome, Stockholm, Madrid, Paris	3600
(GMT+02:00) Athens, Istanbul, Cairo, Helsinki, Jerusalem	7200
(GMT+03:00) Moscow, St. Petersburg	10800
(GMT+05:30) Bombay, Calcutta, Madras, New Delhi	18000
(GMT+08:00) Beijing, Chongqing, Hong Kong, Singapore, Taipei	28800
(GMT+09:00) Osaka, Sapporo, Tokyo, Seoul	32400
(GMT+10:00) Canberra, Melbourne, Sydney	36000

Location	Time zone offset (seconds)
(GMT+12:00) Auckland, Wellington	43200

- **FORCE_TIME_ZONE [YES | NO]** — allows you to force the time zone offset on each user's IP Deskphone. The default is NO.

  - **YES** - forces the IP Deskphone to use the TIMEZONE_OFFSET specified in the device configuration file.
  - **NO** - uses the value stored in the user preferences.

- **IM_MODE [ENCRYPTED | TEXT | SIMPLE | DISABLED]** — is used to configure the mode of Instant Messaging (IM). The default setting is ENCRYPTED.

  - **ENCRYPTED** - Instant Messages are sent encrypted.
  - **TEXT** - Instant Messages are sent as text.
  - **SIMPLE** - Instant Messages are sent using SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) protocol.
  - **DISABLED** - Instant Messaging is turned off and no Instant Messages can be sent or received.

- **IM_NOTIFY [YES | NO]** — is used to turn on or off the Blue LED indicator upon receipt of an Instant Message.

  - **YES** - the Blue LED functions when an Instant Message is received.
  - **NO** - the Blue LED does not function when an Instant Message is received.

  Note:

  If IM_NOTIFY is disabled, the Blue LED continues to operate for other features.

- **DEF_DISPLAY_IM [YES | NO]** — enables or disables the display of Instant Messages (IM). The default setting is NO.

  - **YES** - enables display of IMs.
  - **NO** - disables display of IMs.

- **SELECT_LAST_INCOMING** — is used to determine which call is selected when there are multiple calls ringing (or active). The default value is 0.

  - 0—leaves the last selected call static as new calls come in or are dropped.
  - 1—the selected call in the call list jumps to the most recent ringing call after it is added to the list.

- **MAX_LOGINS [x]** — is used to determine the maximum number of user accounts that can be logged in at the same time. Numbers higher than the number of line keys on the IP Deskphone are equivalent to no limit other than the line keys. A value of 1 allows a single user at a time. A value of 0 is treated the same as a value of 1 because you cannot restrict the IP Deskphone to 0 logins. The number of concurrent logins can never exceed 24, regardless of the value configured on **MAX_LOGINS**. The default is unlimited.

x - the maximum number of user accounts that can be logged in at the same time.

- **MAX_INBOX_ENTRIES [x]** — is used to restrict the maximum number of inbox entries and takes a number as a parameter. For example, **MAX_INBOX_ENTRIES 100** limits the number of entries in the inbox to 100. The default limit is 100.

  x - the maximum number of in box entries.

- **MAX_OUTBOX_ENTRIES [x]** — is used to restrict the maximum number of outbox entries and takes a number as a parameter. For example, **MAX_OUTBOX_ENTRIES 100** limits the number of entries in the outbox to 100. The default limit is 100.

  x - the maximum number of outbox entries.

- **MAX_REJECTREASONS [x]** — is used to restrict the maximum number of Call Decline Reasons (**Prefs > Feature Options > Call Decline Reasons**) and takes a number as a parameter. The default limit is 20.

  x - the maximum number of reject reasons.

- **MAX_CALLSUBJECT [x]** — is used to restrict the maximum number of call subjects (**Prefs > Feature Options > Call Subject**) and takes a number as a parameter. The default limit is 20.

  x - the maximum number of call subject reasons.

- **MAX_PRESENCENOTE [x]** — is used to restrict the maximum number of presence notes and takes a number as a parameter. The default limit is 20.

  x - the maximum number of presence notes that an IP Deskphone can receive.

- **DEF_LANG [language]** — is a command used for configuring the default language. Select one of the supported languages from the language list downloaded. Note that the corresponding language file must be downloaded and stored in the IP Deskphone through the [LANGUAGE] section in Provisioning. If the language file is not stored in the IP Deskphone, the default language English is used.

  language - the default language used.

  Note:

  To update the default language of an IP Deskphone already configured as English to any other language, use the following steps:

  a. Update the Device Configuration file with DEF_LANG configured as the Language file name.

  b. Set FORCE_REBOOT to Yes.

  c. In the 12xxSIP.cfg file, make the Download mode FORCED

  d. Incrementally increase the version number of the 12xxSIP.cfg file.

- **MAX_IM_ENTRIES [x]** — is used to configure the maximum number of Instant Message (IM) entries and takes a number as a parameter. Once the maximum number is reached, the oldest IM is deleted without any user notification. The default limit is 999.

  x - the maximum number of instant messages.

- **MAX_ADDR_BOOK_ENTRIES [x]** — is used to configure the maximum number of entries in the address book and takes a number as a parameter. The default limit is 100.

  x - the maximum number of address book entries.

- **ADDR_BOOK_MODE [NETWORK | LOCAL | BOTH]** — is a command to choose the address book that is used to search for other users. The default setting is NETWORK.

  - **NETWORK** - downloads the user's address book from the network. New address book entries are uploaded to the network.
  - **LOCAL** - creates a user address book and stores it locally on the IP Deskphone.
  - **BOTH** - attempts to download a network address book and keep a copy on the IP Deskphone. If a network address book is available, the IP Deskphone functions as if NETWORK mode has been selected.

- **HOLD_TYPE [RFC2543 | RFC3261]** — is used to select the protocol to hold a call. The default setting is RFC3261.

  - **RFC2543** - RFC2543 is a standard protocol of the Internet Engineering Task Force (IETF).
  - **RFC3261** - RFC3261 is a standard protocol of the IETF.

- **ENABLE_3WAY_CALL [YES | NO]** — is a flag to enable or disable local IP Deskphone-based three-way calling for three-party conferences.

  - **YES** - enables local (IP Deskphone-based) three-way calling for three-party conferences. YES is the default.
  - **NO** - disables local (IP Deskphone-based) three-way calling.

- **DISABLE_PRIVACY_UI [YES | NO]** — is a flag to disable the privacy setting in UI menus. Disabling the privacy setting in UI menus disables the user's ability to configure privacy options.

  - **YES** - disables the privacy setting in the UI menus.
  - **NO** - enables the privacy setting in the UI menus. NO is the default.

DISABLE_OCT_ENDDIAL [YES | NO] — is a flag used to configure the pound (#) key. The default setting is YES.

- **YES** - the pound (#) key initiates dialing when pressed after a phone number is entered.

- **NO** - the pound (#) key functions as any other digit or character on the dial pad typically used in networks that use vertical service codes or access codes.

FORCE_OCT_ENDDIAL [YES | NO] — is a flag used to override attempts to change the function of the pound (#) key on the Graphical User Interface (GUI). The default setting is NO.

- **YES** - overrides attempts to change the function of the pound (#) key on the GUI.
- **NO** - does not override a change of the function of the pound (#) key on the GUI.

SNTP_ENABLE [YES | NO] — allows the IP Deskphone to obtain the time and date from an NTP server. The default is NO.

The IP Deskphone updates the time once every 24 hours from the NTP server. If the IP Deskphone cannot contact the server, the IP Deskphone tries every 15 minutes up to a maximum of 6 attempts, and then hourly attempts are made. If SNTP_ENABLE is configured as NO, the IP Deskphone tries to retrieve the time and date from the SIP proxy server. However, not all SIP proxy servers support this method of retrieving the time and date.

- **YES** - enables NTP.
- **NO** - disables NTP.

SNTP_SERVER [ip_address] — is the IP address or FQDN of the NTP server that provides the time and date to the IP Deskphone. If this is not specified, the IP Deskphone does not generate any NTP requests.

ip_address - the IP address of the NTP server in either Fully Qualified Domain Name (FQDN) or non-FQDN format.

- **MADN_TIMER [x]** — is used to configure the MADN polling timer interval (the interval at which the IP Deskphone attempts to determine the MADN group of the logged-in user). The minimum value for the polling interval is 900 seconds (15 minutes). The default value is 1800.

  x - the time delay (in seconds) between queries to find the MADN group DN of a user. The minimum value 900.

- **MADN_DIALOG [YES |NO]** — is used to configure the SIP URI or the GROUP DN for the subscription to the dialog event. The default value is NO.

  - **YES** - subscribes to the dialog event using the SIP URI of the user.
  - **NO** - subscribes to the dialog event using the group of the user.

- **DEFAULT_CFWD_NOTIFY [YES | NO]** — is used to configure the "ring splash" which occurs when either local call forwarding or network-based call forwarding have been enabled. If this configuration value is enabled, the IP Deskphone plays an abbreviated ring tone to remind the user that a call has been forwarded. This configuration value only effects users when their user profile is first created, unless the FORCE_CFWD_NOTIFY flag is also used. The default setting is NO

  - **YES** - a brief ring splash plays when a call is forwarded.

- **NO** - the ring splash does not play.

- **FORCE_CFWD_NOTIFY [YES | NO]** — allows the administrator to force the behavior of the DEFAULT_CFWD_NOTIFY value on all users who login to the IP Deskphone. The default setting is NO.

  - **YES** - the DEFAULT_CFWD_NOTIFY configuration value is forced into effect for the user.

  - **NO** - the configuration value is not forced into effect for the user.

- **ENABLE_SERVICE_PACKAGE [YES | NO]** — toggles the subscription to the Call Server service package. When the IP Deskphone connects to a Call Server that does not recognize the service package, the subscription for the service package fails. If this happens, ad hoc conferencing is not available, even if the Call Server supports ad hoc conferencing. You can configure values for ad hoc conferencing when the service package is not retrieved. The IP Deskphone retrieves the service package based on a configurable Boolean value.

  - **YES** - the IP Deskphone downloads the service package.

  - **NO** - the IP Deskphone does not download the service package.

  CONFERENCE_URI1 — contains the conference Uniform Resource Identifier (URI); for example CONFERENCE_URI1 conference@bvw.com.

  ADHOC_ENABLED1 [YES | NO]

  - **YES** - the Call Server supports ad hoc conferencing.

  - **NO** - the Call Server does not support ad hoc conferencing.

  MAX_ADHOC_PORTS1 [0–4] — indicates the maximum number of users supported for ad hoc conferencing on the server. This value must be the same as the value configured on the server. When ENABLE_SERVICE_PACKAGE is enabled, the preceding parameters are ignored.

- **INTERCOM_PAGING [YES | NO]** — allows the IP Deskphone to belong to a paging group. When a page group call is received, a one-way speech path is created to the IP Deskphone, and the IP Deskphone automatically goes to a hands-free intercom state.

  - **YES** - intercom/paging functionality is enabled.

  - **NO** - intercom/paging functionality is disabled.

- **LOGOUT_WITHOUT_PASSWORD [YES | NO]** — allows the user to log off without entering their password if the administrator enables the LOGOUT_WITHOUT_PASSWORD feature.

  - **YES** - enables the user to logout without a password.

  - **NO** - does not allow the user to logout without a password.

- **REMOTE_CHECK_FOR_UPDATE [YES | NO]** — provides the functionality to start a check to remotely update the IP Deskphone with the latest firmware present on the Trivial

File Transfer Protocol (TFTP) server. You can enable or disable this feature with the flag present in the Device Configuration file. By default, this flag is set to NO.

- **YES** - prompt the user about the scheduled event and the user can accept or reject the scheduled firmware update check by pressing the **YES** or **NO** soft key.
- **NO** - disables the remote check for update option.

- **SECURE _INCALL_DIGITS [YES | NO]** — shows the typed digits as asterisks when the user makes a call into voice mail. When this feature is enabled, the most recently-pressed key is displayed but is overwritten by an asterisk (*) when the next key is pressed. The user has the option to Hide or Unhide the digits typed.

  - **YES** - provides the secure digits while in call functionality.
  - **NO** - disables the secure digits while in call functionality.

- **E911_USERNAME [username]** — is an emergency username used for making an emergency call that does not require login. The proxy must be configured with the same emergency username, otherwise, the emergency call fails.
- **E911_PROXY [proxy_name]** — is a default emergency proxy. This variable must contain the value that matches the value defined by one of the following variables specified in the same config file:

  - SIP_DOMAIN1
  - SIP_DOMAIN2
  - SIP_DOMAIN3
  - SIP_DOMAIN4
  - SIP_DOMAIN5

  If E911_PROXY does not match the value defined by these five variables, or the variable E911_PROXY is not defined, the value of SIP_DOMAIN1 is used as the emergency proxy.

- **E911_PASSWORD [password]** — is the password for the emergency username that is used for making an emergency call that does not require login. The proxy must be configured with the same password; otherwise the emergency call fails.
- **E911_TXLOC [Register | Invite]** — is the variable that describes location information that must be sent with the REGISTER SIP message, or with the INVITE SIP message.

  - **REGISTER** – the location is sent in both the INVITE and the REGISTER message.
  - **INVITE** – the location is sent with the INVITE only.

- **MENU_AUTO_BACKOUT [x]** — is a menu auto back-out time preference configuration used to configure the auto back-out time on newly-created profiles (not for profiles that already exist). The values, in seconds, are 0, 15, 30, 60, 120, 300, 600. The default value is 30.

Example:

MENU_AUTO_BACKOUT 15

Note:

There are some application screens that do not time out. Some menus, such as the administration menus, require the user to press the Back or Quit key to exit the screen.

- **AUTOCLEAR_NEWCALL_MSG** —is used to configure the missed calls notification mode. Y means that the notification is cleared as soon as the inbox is entered (without needing to visit all missed entries. The values are Y and N. The default value is N.

  This configuration value only affects users when a user profile is first created. It does not affect a user profile which already exists. A user can modify the feature parameter by using the **Preferences** menu on the IP Deskphone and then selecting the **Feature Options > Missed Call Notification** menu item.

- **LOGIN_BANNER_ENABLE** — is used to enable or disable the customizable login banner. If configured as enable, the flag causes the login of the primary user to display the provisioned banner text as part of the login process. The banner text file is a separate file downloaded by provisioning. The banner text file is specified much like the current dialing plan is specified (file name listed in 12xxSIP.cfg, under section [LOGIN_BANNER]), and is downloaded when enabled or disabled. To be accepted, the file must contain at least one byte and must be no bigger than 2048 bytes. The encoding of the file must be UTF-8, or compatible with UTF-8, to ensure that all the characters are displayed properly. The values are Y and N. The default value is N.

- **SECURE_UI_ENABLE** — is used to disable access to the IP Deskphone Information details screen, and the context-sensitive soft key that invokes it. The values are YES and NO. The default value is NO.

  - **YES**— disables access to the IP Deskphone Information details screen and the context-sensitive soft key that invokes it.
  - **NO**—enables access to the IP Deskphone Information details screen and the context-sensitive soft key that invokes it.

- **BG_IMAGE_ENABLE** — is used to configure the background image file for the display in newly created profiles, and can completely disable the background image feature and disable the corresponding user interface. If the specified file does not exist in the images folder of the IP Deskphone, no background image is used for the display. The values are Y and N. The default value is Y.

  Configuration flag:

  BG_IMAGE_ENABLE Y/N (default: Y)

- **BG_IMG_SELECT_ENABLE** — is used to change the selected background image for the display. If the flag is configured to N, the UI to change the background image is hidden from the user, locking the currently configured image as the background image on the IP Deskphone. The values are Y and N. The default value is Y.

Configuration flag:

BM_IMG_SELECT_ENABLE Y/N (default: Y)

- **USE_BG_IMAGE** — is used to configure the background image for the display of newly created profiles by specifying a file name available on the FFS. BG_IMAGE_ENABLE must be configured as Y in order to select a background image.

  Configuration flag:

  USE_BG_IMAGE <Image filename>

  Note:
  Image files for the IP Deskphone must include the PNG format.

- **SPEEDLIST_KEY_INDEX** — is used to specify the programmable key used for displaying the Speed Dial List. If the specified index does not exist on the IP Deskphone, or is invalid, the speed dial list is not displayed on the IP Deskphone. The IP Deskphone retrieves the device configuration through provisioning. If the SPEEDLIST_KEY_INDEX flag is configured to a valid programmable key that can be used for the feature, for example, >1 and less than or equal to available number of programmable keys, the IP Deskphone verifies if it has previously loaded a "Speed Dial List" file (a file containing the contents of the speed dial list). This file is similar to the dialing plan file. It needs to be properly configured and uploaded to the IP Deskphone through provisioning. The IP Deskphone parses the file, and configures the feature key specified by SPEEDLIST_KEY_INDEX to hold the Speed Dial List. If the key defined for use by the Speed Dial List is already in use, the key is overwritten and the key is assigned speed dial list functionality. The Speed Dial List feature key then uses the label that is provisioned in SPEEDLIST_LABEL which cannot be modified by the end user.

  Configuration flag::

  SPEEDLIST_KEY_INDEX <feature key index>

- **SPEEDLIST_LABEL** — is a feature key label used by the speed dial list feature key. The default value is SDL.

  Configuration flag:

  <SPEEDLIST_LABEL <text>

- **BLF_ENABLE** — is used to enable or disable the Busy Lamp Field (BLF) feature support. If configured as Y, the flag BLF_RESOURCE_LIST_URI is not ignored and the BLF feature is used. The values are Y, N, SCS, and SIPX. The default is N.

  When BLF_ENABLE has the SCS or SIPX value, the BLF_RESOURCE_LIST_URI parameter is ignored and the IP Deskphone autogenerates an URI of the following format: `~~rl~C~<username>@<domain>`

- **BLF_RESOURCE_LIST_URI** — is used to configure the Busy Lamp Field (BLF) resource list URI for the BLF feature. You must use the URI provided by the proxy when properly configuring the user for BLF.

Configuration flag:

```
BLF_RESOURCE_LIST_URI <blf uri>
```

The `<blf uri>` is the server provided URI to subscribe for BLF notifications, for example, `blf-resource-list@as.avaya.com` .

- **FM_SOUNDS_ENABLE** — allows the user to act on WAV files using the file manager. If the value is configured as N, the IP Deskphone cannot perform any actions on WAV files, such as delete or copy a wav file, through the file manager. If the user selects a WAV file on the IP Deskphone and presses the Delete or Send Context-sensitive softkey, an error message appears. If the value is configured as Y, the user can delete or copy WAV files with the file manager interface (this applies to WAV files on the IP Deskphone). The values are Y and N. The default value is Y.

  - **YES**—allows the user to delete or copy WAV files on the IP Deskphone through the file manager
  - **NO**—does not allow the user to delete of copy WAV files on the IP Deskphone through the file manager

  Configuration flag:

  FM_SOUNDS_ENABLE Y/N (default: Y)

- **FM_IMAGES_ENABLE** — allows the user to act on JPG and PNG files using the file manager. The values are Y and N. The default value is Y.

  Configuration flag:

  FM_IMAGES_ENABLE Y/N (default: Y)

- **FM_CERTS_ENABLE** — allows the user to act on CER and PEM files using the file manager. The values are Y and N. The default value is N.

  Configuration flag:

  FM_CERTS_ENABLE Y/N (default: N)

- **FM_CONFIG_ENABLE** — allows the user to act on CFG files using the file manager. The values are Y and N. The default value is N.

  Configuration flag:

  FM_CONFIG_ENABLE Y/N (default: N)

- **FM_LOGS_ENABLE** — allows the user to act on CFG files using the file manager. The values are Y and N. The default value is Y.

  Configuration flag:

  FM_LOGS_ENABLE Y/N (default: Y)

- **HOTLINE_ENABLE** — indicates if Hotline Service is enabled or disabled. The values are Yes and No. The default value is No.

- **HOTLINE_URL** — is used as To field of INVITE message by the SIP IP Deskphone to notify the Proxy Server that this is a call from a Hotline Deskphone. The HOTLINE_URL is not a real URL of the Hotline target. The IP Deskphone has no idea about the Hotline target. The Proxy server replaces the To field of INVITE request message with a real Hotline target when it receives an INVITE request from the Hotline Phone. The default value is Hotline.
- **SESSION_TIMER_ENABLE** — indicates if the session timer service is enabled or disabled. The values are Yes and No. The default value is Yes.
    - **YES**—the Session Timer Service for the IP Deskphone is enabled, and the behavior of the IP Deskphone complies with RFC4028.
    - **NO**—the Session Timer Service is disabled.
-
- **SESSION_TIMER_MIN_SE** — indicates the minimum session expiration in seconds. The default value is 1800.
- **SET_REQ_REFRESHER** — indicates what refresher value is configured in the initial session request. The values are 0, 1, and 2. The default value is 0.
    - 0—indicates that the refresher is omitted
    - 1—indicates that the refresher is configured to UAC
    - 2—indicates that the refresher is configured to UAS
- **SET_RESP_REFRESHER** — indicates what refresher value is configured in the 200 OK response. The values are 0, 1, and 2. The default value is 2.
    - 0—indicates that the refresher is omitted (only valid when SET_REQ_REFRESHER is not equal to 0)
    - 1—indicates that the refresher is configured to UAS
    - 2—indicates that the refresher is configured to UAC
- **ENABLE_INTERWORKING** — is used to enable the interworking feature to pre-authorize users or groups of users to access automatic call answer. The configuration values are YES and NO. The default value is NO.
    - **YES**—the interworking feature is enabled.
    - **NO**—the interworking feature is disabled.
- **MAX_ALLOWEDADDRESSES** — limits the size of the list of user and domain addresses stored for auto-answered authorization. The default value is 100.
- **PORT_MIRROR_ENABLE** — is used to enable or disable the Port Mirroring feature. The values are YES and NO. The default value is NO.
    - **YES**—The Port Mirroring prompt in the Advanced Diag Tools dialog is enabled and can be modified.

  - **NO**—The Port Mirroring prompt in the Advanced Diag Tools dialog is disabled (dimmed) and cannot be modified.

- **MEMCHECK_PERIOD** — is used to determine the time period in seconds when the Memory monitor wakes up (after re-start or the last memory check attempt). The values are 1800 (0.5 hrs) to 86400 (24 hrs). The default value is 86400 (24 hrs).
- **DOS_PACKET_RATE** — determines the maximum number of packets per second that is allowed.
- **DOS_MAX_LIMIT** — specifies how many packets past the DOS_PACKET_RATE the IP Deskphone can receive before packets are dropped. If packets are received at a rate of DOS_PACKET_RATE +1, then packets are dropped after the time specified in DOS_MAX_LIMIT (in seconds).
- **DOS_LOCK_TIME** — specifies the amount of time (in seconds) that the IP Deskphone stops processing packets after DOS_MAX_LIMIT is reached. If DOS_PACKET_RATE is < 1, other values are ignored and packets are not dropped.
- **LOGSIP_ENABLE** — is used to enable or disable SIP-logging. The values are YES and NO. The default value is NO.
  - **YES**—the SIP-logging Manager is active and starts to log SIP incoming and outgoing packages into the log file in FFS.
  - **NO**—the SIP-logging Manager is not active and cannot log SIP incoming and outgoing packages into the log file in FFS.
- **CUST_CERT_ACCEPT** — is a Security Policy parameter that controls further signing of a customer root certificate (not the first one).

  The values are VAL_NO_MANUAL, VAL_MANUAL_A, and VAL_MANUAL_B. The default value is VAL_MANUAL_A.
- **SEC_POLICY_ACCEPT** —allows you to access the Certificate Administration User Interface. The values are YES and NO. The default value is NO.
- allows you to accept security policy. The default value is VAL_MANUAL_A. Following are the acceptable parameters:
  - **VAL_MANUAL_A**—If the resource file is not signed and if there are no customer certificates, then Finger Print Display and Accept/Reject options appear .
  - **VAL_MANUAL_B**—If the resource file is not signed and if there are no customer certificates, enter the Finger Print Value manually and then select Accept option.
- **SECURITY_LOG_UI_ENABLE** — allows you to access the Security and Error Logs User Interface. The values are YES and NO. The default value is No.
- **KEY_SIZE** — is the default key size that is used when generating keys on the IP Deskphone, and acts at the minimum allowed key size that is enforced when loading certificates from the IP Deskphone. The values are 1024, 1536, and 2048. The default value is 1024.

- **KEY_ALGORITHM** — is the preferred key generation algorithm. The accepted value is KEY_ALG_RSA.
- **TLS_CIPHER** — is the preferred TLS Cipher used for HTTPS to configure a stronger cipher preference when available. The values are RSA_WITH_AES_128_CBC_SHA, and RSA_WITH_AES_256_CBC_SHA. The default value is RSA_WITH_AES_256_CBC_SHA.
- **SIGN_SIP_CONFIG_FILES** — overrides the file signing of files (resource files such as the device configuration file and the dial plan) other than the Security Policy and Customer Certificates. The values are YES and NO. The default value is NO.
    - **YES**—Signing is required.
    - **NO**—No authentication check is performed.
- **FP_PRESENTED** — allows you to accept or reject a Finger Print if the resource file is not signed and if there are no customer certificates.
- **FP_ENTERED** — allows you to manually enter and accept a Finger Print value if the resource file is not signed and if there are no customer certificates.
- **SUBJ_ALT_NAME_CHECK_ENABLE** — allows you to verify the Subject Alternative Attribute in the presented certificate. Only the IPv4 IP address is supported for this attribute. The values are YES and NO. The default value is NO.
- **SECURITY_POLICY_PARAM_CHANGE** — allows the IP Deskphone to enter changes that are made to the security policy file in the security log file.
- **CERT_EXPIRE** — allows you to select Certificate Expiration Policy. The default value is LOG_EXPIRE. Following are the acceptable parameter values:
    - **DELETE_CERT**—A certificate is deleted when it expires and a security log entry is added.
    - **LOG_EXPIRE**—A certificate is not deleted when it expires and a security log entry is added. Even if the certificate is not deleted, it cannot be used to authenticate a file.
    - **NO_EXPIRE_LOG**—A certificate is not deleted when it expires and security log entry is not added. Even if the certificate is not deleted, it cannot be used to authenticate a file.
- **DWNLD_CFG_ACCEPT** — defines how all TFTP configuration files are authenticated when there are no customer certificates on the phone. The parameter does not come to effect when a customer certificate installed. The default value of the parameter is VAL_ACCEPT.

  Following are the acceptable parameter values:

    - **VAL_ACCEPT**—Unsigned and signed files are always accepted if there are no valid customer certificates.
    - **VAL_MANUAL_A**—If the resource file is not signed and if there are no customer certificates, then Finger Print Display and Accept/Reject options prompt appears.

  - **VAL_MANUAL_B**—If the resource file is not signed and if there are no customer certificates, then enter Finger Print Value and select the Accept option manually.

- **DWNLD_CFG_SIGNING** — defines if configuration files (12xxSIP.cfg) are forced to sign if a customer certificate installed. This parameter does not come to effect if the customer certificates are installed. The default parameter value is no. The following are the acceptable values for this parameter:

  - **NO** — If there is a customer certificate installed, the downloaded file is automatically accepted without authentication.

  - **YES** — If there is a customer certificate installed, the downloaded file must be signed and fully authenticated.

QoS and ToS commands

- AVAYA_AUTOMATIC_QoS [YES | NO] provides a better treatment for signaling and media packets after you deploy the IP Deskphones with the Avaya switches. All the devices use private Differentiated Services Code Point (DSCP) values to give better treatment to the traffic coming from peer Avaya devices.

  - YES - the IP Deskphone uses private DSCP values, unless overridden.

  - NO - the IP Deskphone uses either one of the configured DSCP values or the system default values.

- DSCP_CONTROL [x] is a value entered in decimal format between -1 and 63. If the value is -1, the DSCP value is picked up by the Service Package. The default value is 40.

  x - a value from -1 to 63 indicating the DSCP value.

- 802.1P_CONTROL [x] is a value entered in decimal format between -1 and 7 representing the 802.1P value in the SIP signaling packets. If the value is -1, the 802.1P value is retrieved from the Service Package. The default value is 6.

  x - the value from -1 to 7 indicating the 802.1P value.

- DSCP_MEDIA [x] is a value entered in decimal format between -1 and 63 representing the DSCP value in the Real-time Transfer Protocol packets. If the value is -1, the DSCP value is retrieved from the Service Package. The default value is 44.

  x - a value from -1 to 63 indicating the DSCP value.

- 802.1P_MEDIA [x] is a value entered in decimal format between -1 and 7 representing the 802.1P value in the IP Deskphone Media (RTP) packets. If the value is -1 then the 802.1P value is retrieved from the Service Package is the 802.1 setting for media Real-time Transport Protocol (RTP). The default value is -1.

  x - a value from -1 to 7 indicating the 802.1P value.

- DSCP_DATA [x] is a value entered in decimal format between -1 and 63 representing the DSCP value in the provisioning packets. If the value is -1, the DSCP value is retrieved from the Service Package. The default value is 40.

  x - a value from -1 to 63 indicating the DSCP value.
- 802.1P_DATA [x] is a value entered in decimal format between -1 and 7 representing the 802.1P value in the provisioning packets. If the value is -1, the 802.1P value is retrieved from the Service Package. The default value is 6.

  x - a value from -1 to 7 indicating the 802.1P value.

Tone configuration commands

- DIAL_TONE [frequency1 | frequency2 | on_time | off_time] is used to select the tone advising the caller that the exchange is ready to receive call information and invites the user to start sending call information. You can select the country-specific tone. The default tone is the North American tone.

  - frequency1 - the frequency of tone 1.

  - frequency2 - the frequency of tone 2.

  - on_time - the duration of the tone when it is on. A -1 indicates a continuous tone.

  - off_time - the duration when no tone is played.

  The following is an example of DIAL_TONE:

  350,440;-1 (350 and 440 Hz continuous tone)
- RINGING_TONE [frequency1 | frequency2 | on_time | off_time] is used to select the tone advising the caller that a connection is made and a calling signal is applied to a telephone number or service point. You can select the country-specific tone. The default tone is the North American tone.

  - frequency1 - the frequency of tone 1.

  - frequency2 - the frequency of tone 2.

  - on_time - the duration of the tone when it is on. A -1 indicates a continuous tone.

  - off_time - the duration when no tone is played.

  The following is an example of RINGING_TONE:

  440,480; 2000,4000 (440 and 480 Hz with 2 seconds on, 4 seconds off)

- BUSY_TONE [frequency1 | frequency2 | on_time | off_time] is used to select the tone advising the caller that the telephone number is busy. You can select the country-specific tone. The default tone is the North American tone.
  - frequency1 - the frequency of tone 1.
  - frequency2 - the frequency of tone 2.
  - on_time - the duration of the tone when it is on. A -1 indicates a continuous tone.
  - off_time - the duration when no tone is played.
- FASTBUSY_TONE [frequency1 | frequency2 | on_time | off_time] is used to select the tone advising the caller that the telephone number is busy. It is fast in cadence or frequency. You can select the country-specific tone. The default tone is the North American tone.
  - frequency1 - the frequency of tone 1.
  - frequency2 - the frequency of tone 2.
  - on_time - the duration of the tone when it is on. A -1 indicates a continuous tone.
  - off_time - the duration when no tone is played.
- CONGESTION_TONE [frequency1 | frequency2 | on_time | off_time] is used to select the tone advising the caller that the groups of lines or switching equipment necessary for setting up the required call, or for the use of a specific service, are temporarily engaged. You can select the country-specific tone. The default tone is the North American tone.
  - frequency1 - the frequency of tone 1.
  - frequency2 - the frequency of tone 2.
  - on_time - the duration of the tone when it is on. A -1 indicates a continuous tone.
  - off_time - the duration when no tone is played.

  The IP Deskphone supports using WAV files to replace the ringtone Frequency/Cadence pattern. For a system-wide setting, the country default values can be used.

NAT configuration commands

- NAT_SIGNALLING [NONE | SIP_PING | STUN] indicates the type of protocol used for NAT traversal in the signaling port. The IP Deskphone with SIP Software supports two methods of NAT traversal of the signaling path: SIP_PING and STUN.
  - NONE - If the value is not configured as None, this parameter overrides the value of the parameter SIP_PING in the device configuration file.
  - SIP_PING - a Avaya proprietary NAT traversal protocol. Note that SIP_PING only supports NAT traversal in the signaling port.

STUN - the most common NAT traversal method.

- NAT_MEDIA [NONE | STUN] indicates the type of protocol used for NAT traversal in the media ports. The default is NONE.
  - NONE - is the default and disables NAT_MEDIA.
  - STUN - the most common NAT traversal protocol for the media (RTP and Real-time Control Protocol [RTCP]) port.
  - x - is the binding lifetime in seconds.

> **Important:**
> NAT_TTL [x] is used for future development. Currently, the default value is 2 minutes (120 seconds) and IP Deskphones does not process or use the value defined in NAT_TTL [x]. The IP Deskphones always pings the ports at regular intervals of 60 seconds regardless of the NAT_TTL value.

- STUN_SERVER_IP1[ip_address] NAT traversal using STUN protocol requires a STUN server in the public internet. Two STUN server IPs can be provisioned.

  ip_address - is the IP address of STUN server 1.
- STUN_SERVER IP2[ip_address] NAT traversal using STUN protocol requires a STUN server in the public internet. Two STUN servers IPs can be provisioned.

  ip_address - is the IP address of STUN server 2.
- STUN_SERVER_PORT1[port_number] is the port number used corresponding to STUN_SERVER_IP1. The default port number is 3478.

  port_number - is the port number.
- STUN_SERVER_PORT2[port_number] is the port number used corresponding to STUN_SERVER_IP2. The default port number is 3478.

  port_number - is the port number.

VQMon configuration commands

It is important to read How VQMon works on page 98 before configuring the VQMON parameters.

- VQMON_PUBLISH [YES | NO] is the command that is used to enable or disable the publish message containing the voice quality monitoring metrics sent to the Proactive Voice Quality Monitoring (PVQMoN) collecting server.
  - YES - enables VQMoN.

- NO - disables VQMoN. NO is the default.

- VQMON_PUBLISH_IP [xxx.xxx.xxx.xxx] is used to configure the IP address of the PVQMoN server that collects voice quality monitoring metrics from the publish message.

  This IP address is used only within the report.

- LISTENING_R_ENABLE [YES | NO] is used to enable or disable the alerts based on the Listening R Minor and Major Thresholds. The default value is vocoder-dependent, using a scale from 1 (lowest quality) to 100 (highest quality). Currently, default values are used based on VOCODER on a per-call basis as summarized below.

  - YES - enables the sending of the alert report based on the Listening R Value.

  - NO - disables the sending of the alert report based on the Listening R Value.

VOCODER_G711_ULAW
VOCODER_G711_ULAWPLP
	LISTENING_R_WARN = 80
	LISTENING_R_EXCE = 70

VOCODER_G723
VOCODER_FLAG_G723_RATE_53
VOCODER_FLAG_G723_RATE_63
	LISTENING_R_WARN = 60
	LISTENING_R_EXCE = 50

VOCODER_G729 VOCODER_PCM8 vqmonVocoderTypeUnknown	LISTENING_R_WARN = 70 (default if not configured and unknown type) LISTENING_R_EXCE = 60

- LISTENING_R_WARN [xx] is the threshold to send a report on Listening R less than [xx]. The default value is 70. Using 0 resets it to default based on far end VOCODER.

  xx - is an INTEGER value used as threshold.

- LISTENING_R_EXCE [xx] is the threshold to send a report on Listening R less than [xx]. The default value is 60. Using 0 resets it to default based on far end VOCODER.

  xx - is an INTEGER value used as threshold.

- PACKET_LOSS_ENABLE [YES | NO] is used to enable or disable the alerts based on the packet loss thresholds. Packet loss is the fraction of RTP data packets from the source lost since the beginning of reception. The value is an integer scaled by 256. The range is 1 to 25600.

  - YES - enables the sending of alert report based on the packet loss

  - NO - disables the sending of alert report based on the packet loss

- PACKET_LOSS_WARN [xx] is the threshold to send a report on Packet Loss greater than [xx]. The default is 256 (1%). Using 0 resets the threshold to default.

  xx - is an INTEGER value scaled by 256 that is used as threshold. The range is 1 to 25600.

- PACKET_LOSS_EXCE [xx] is the threshold to send a report on Packet Loss greater than [xx]. The default is 1280 (5%). Using 0 resets the threshold to default.

  xx - is an INTEGER value scaled by 256 that is used as threshold. The range is 1 to 25600.

- JITTER_ENABLE [YES | NO] is used to enable or disable alerts based on the inter-arrival Jitter on incoming RTP packets inter-arrival time. The value is represented in 1/65536 of a second.

  - YES - enables the sending of alert report based on jitter detection

  - NO - disables the sending of alert report based on jitter detection

- JITTER_WARN [xx] is the threshold to send a report on Inter-arrival Jitter greater than [xx]. 1 second is broken up into 65535 (0xffff hex) parts. [xx] / 65535 is the threshold in seconds. The default is 3276 (50 ms). Using 0 resets the threshold to default.

  xx - is an INTEGER value used as threshold

- JITTER_EXCE [xx] is the threshold to send a report on Inter-arrival Jitter greater than [xx]. 1 second is broken up into 65535 (0xffff hex) parts. [xx] / 65535 is the threshold in seconds. The default is 32760 (500 ms). Using 0 resets the threshold to default.

  xx - is an INTEGER value used as threshold

- DELAY_ENABLE [YES | NO] is used to enable or disable the alerts based on the excessive delay detection. This is the one-way delay (including system delay) for the call, measured in milliseconds.

  - YES - enables Excessive delay detection.

  - NO - disables Excessive delay detection.

- DELAY_WARN [xx] is the threshold to give warning on Excessive Delay greater than [xx]. The default is 150 ms. Using 0 resets the threshold to default.

  xx - is an INTEGER value used as a threshold measured in 1/1000 of a second.

- DELAY_EXCE [xx] is the threshold to report unacceptable Excessive Delay greater than [xx]. The default is 175 ms. Using 0 resets the threshold to default.

  xx - is an INTEGER value used as a threshold measured in 1/1000 of a second.

- SESSION_RPT_EN [YES | NO] is used to enable or disable periodic VQMon session reports. The default is disabled.

  Both session report enable and session report interval must be configured if the IP Deskphone software has been upgraded to SIP Release 3.0. Otherwise, the SESSION_RPT_INT default of 60 seconds is used automatically.

  - YES - enables periodic VQMon session reports.

- NO - disables periodic VQMon session reports. Default is NO.

- SESSION_RPT_INT [xx] is used to specify the interval for the periodic VQMon session report in seconds. The minimum acceptable value is 60 seconds. The maximum acceptable value is 600 seconds. The default is 60 seconds.

  xx - is an INTEGER value in seconds.

System commands

ADMIN_PASSWORD [password] is used to change the default administrator password of the IP Deskphone that is used for unlocking network menus. The default is 26567*738.

— password - the administrator password.

IP Deskphone recovery command

RECOVERY_LEVEL controls the IP Deskphone recovery if the IP Deskphone hits any Major or Critical error. The following values are used for setting the recovery level on the IP Deskphone:

- 0 - IP Deskphone never recovers from any error
- 1 - IP Deskphone recovers from Major error
- 2 - IP Deskphone recovers from Major and Critical errors

Default is 255, which is equivalent to the recovery level of 2.

-

-

-

Create the IP Deskphone-specific configuration file on the provisioning server

If the IP Deskphone encounters a [USER_CONFIG] section while parsing the 12xxSIP.cfg configuration file, the IP Deskphone downloads the IP Deskphone-specific configuration file SIP<mac id>.cfg.

IP Deskphone-specific configuration files support customizing the IP Deskphone on a IP Deskphone/user level. Parameters in the device configuration file can be overwritten with a IP Deskphone-specific configuration file.

Most of the parameters in the IP Deskphone configuration file are saved on the IP Deskphone. Removing a parameter from the IP Deskphone configuration file does not change the parameters saved on a configured IP Deskphone. If a parameter is configured only in the IP Deskphone-specific configuration file, removing the IP Deskphone-specific configuration file does not clear the setting.

Important:

If the 12xxSIP.cfg configuration file contains a [USER_CONFIG] section, Avaya recommends that DOWNLOAD_MODE be configured as FORCED. This is a global setting for all IP Deskphones used to determine if the mac id file should be read. Alternatively, if the user wants to use DOWNLOAD_MODE configured to AUTO, when a change is made to any mac id file the version number should be incremented so that all IP Deskphones read the file.

Table 7: IP Deskphone configuration commands on page 70 provides a summary of the commands that can be used in the IP Deskphone configuration file. The syntax of each command is summarized in IP Deskphone configuration commands summary on page 70.

Table 7: IP Deskphone configuration commands

Auto login	AUTOLOGIN_ID_KEY01 AUTOLOGIN_PASSWD_KEY01

IP Deskphone configuration commands summary

- **AUTOLOGIN_ID_KEY01 [* |userID@domain name]**

  This parameter is located within the IP Deskphone-specific configuration file. This is the ID the IP Deskphone uses to register and authenticate. The default User ID "user1" is used if an ID is not supplied and the IP Deskphone is not logged in.

  - ***** - indicates that the IP Deskphone should use its mac address (lower case) as the User ID
  - **userID@domain name** - the user ID must be followed by the domain name; for example, jsmith@mycompany.com; 2247@avaya.com

  Note:

  To provision AUTOLOGIN_ID_KEY01, the IP Deskphone must be rebooted after the IP Deskphone configuration file is updated. To force a hard reboot after the

IP Deskphone configuration file is updated, configure FORCE_REBOOT YES in the device configuration file.

- **AUTOLOGIN_PASSWD_KEY01**

  This parameter is located within the IP Deskphone-specific configuration file. There is no default password. If this is blank and AUTOLOGIN_ENABLE is configured to USE_AUTOLOGIN_ID (or 2) in the device configuration file, the IP Deskphone does not log on.

  Note:

  To provision AUTOLOGIN_PASSWD_KEY01, the IP Deskphone must be rebooted after the IP Deskphone configuration file is updated. To force a hard reboot after the IP Deskphone configuration file is updated, configure FORCE_REBOOT YES in the device configuration file.

Create the Dialing Plan file on the provisioning server

A dialing plan essentially describes the number and pattern of digits that a user dials to reach a particular telephone number. Access codes, area codes, specialized codes, and combinations of the number of digits dialed are all part of a dialing plan.

The purpose of the dialing plan is so that the end user does not have to press the send or pound key (#) to have the IP Deskphone with SIP Software send the initial message to start the call.

Dialing a telephone number on an IP Deskphone that supports SIP can be different than dialing a number from a traditional telephone. SIP signaling is communicated through a SIP URI to get to the far end. For example, you can key in the SIP address, jsmith@yourcompany.com to reach John Smith. When the IP Deskphone with SIP Software receives this address, the dialing plan is bypassed and the IP Deskphone uses the SIP URI to send a SIP INVITE to jsmith@yourcompany.com (INVITE sip: jsmith@yourcompany.com).

Entering a SIP URI address, however, is inconvenient for an IP Deskphone with SIP Software. Also, the user must explicitly press the send key (or use some method to indicate the end of the URI) to indicate the completion of the SIP address. This is not something that the user is accustomed to in a traditional PBX environment.

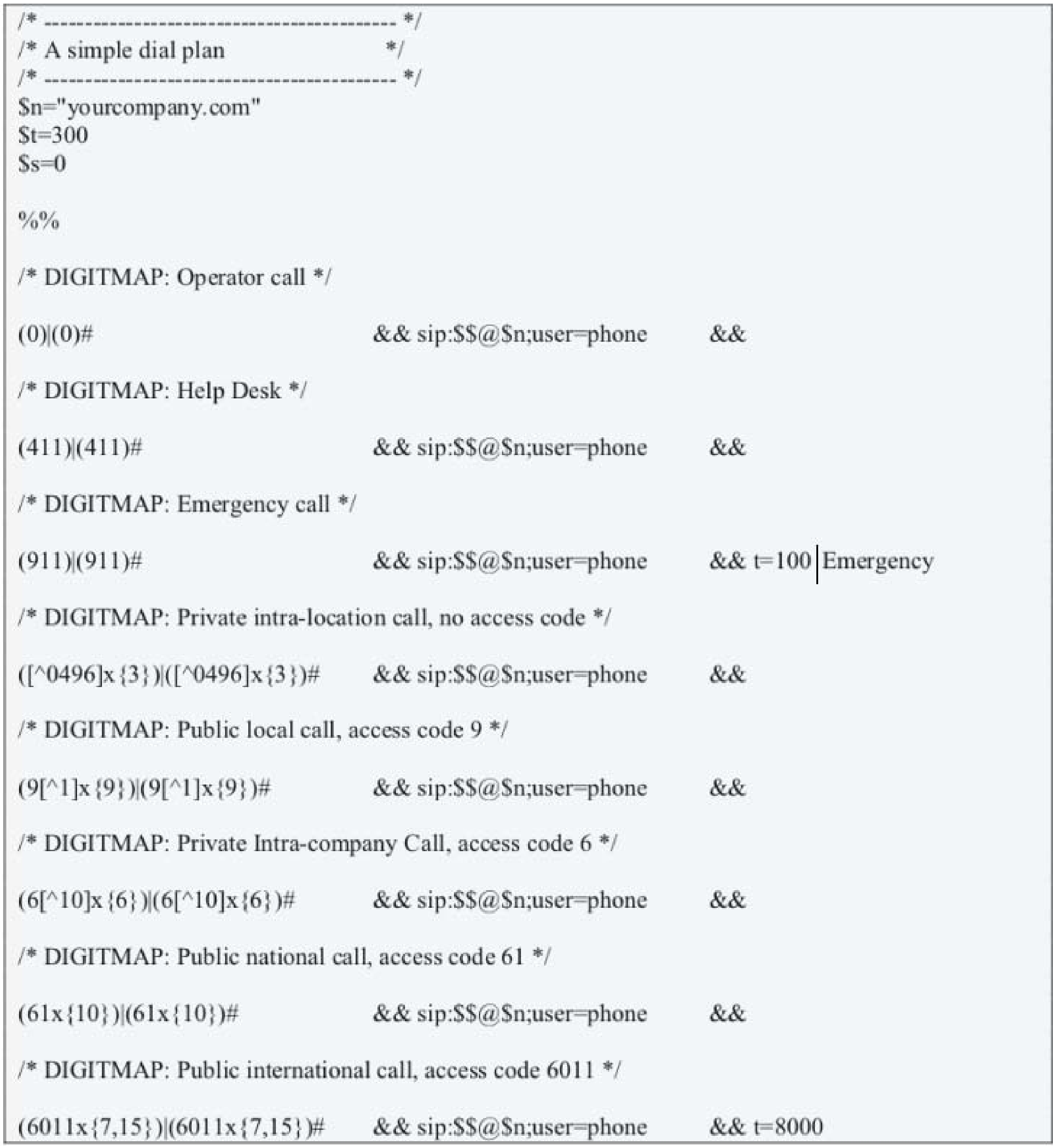
The alternative is to use a URI where numbers are used to reach the far end. Using different access codes, the IP Deskphone with SIP Software translates the digits entered into something that the server can understand and remaps the number entered into different URIs. Some of the numbers are mapped as intercom calls, some numbers are mapped as local Public Switched Telephone Network (PSTN) calls, and some numbers are mapped as public long-distance calls.

The issue is that until the IP Deskphone itself can determine the type of call, no SIP INVITE message is sent. This is where the dialing plan comes into effect. The call type is determined by the dialing plan. Based on the rules defined in the dialing plan, once a match has been identified, the IP Deskphone with SIP Software sends the invite without the need to press the send key. This behavior closely matches the traditional PBX operation.

The IP Deskphone with SIP Software design places no restriction in the format of the SIP URI. The dialing plan is a scheme to match the user experience with traditional PBX operation. It does not restrict the type of URI that the user can use.

The IP Deskphone with SIP Software uses a dialing plan to recognize a call as an call when it sends an INVITE. The dialing plan can have multiple emergency numbers. See the chapter Emergency Services on page 165 for information on the handling of Emergency calls by the IP Deskphone with SIP software.

The following is an example of a dialing plan.



```
/* ----------------------------------------- */
/* A simple dial plan                        */
/* ----------------------------------------- */
$n="yourcompany.com"
$t=300
$s=0

%%

/* DIGITMAP: Operator call */

(0)|(0)#                          && sip:$$@$n;user=phone       &&

/* DIGITMAP: Help Desk */

(411)|(411)#                      && sip:$$@$n;user=phone       &&

/* DIGITMAP: Emergency call */

(911)|(911)#                      && sip:$$@$n;user=phone       && t=100 Emergency

/* DIGITMAP: Private intra-location call, no access code */

([^0496]x{3})|([^0496]x{3})#      && sip:$$@$n;user=phone       &&

/* DIGITMAP: Public local call, access code 9 */

(9[^1]x{9})|(9[^1]x{9})#          && sip:$$@$n;user=phone       &&

/* DIGITMAP: Private Intra-company Call, access code 6 */

(6[^10]x{6})|(6[^10]x{6})#        && sip:$$@$n;user=phone       &&

/* DIGITMAP: Public national call, access code 61 */

(61x{10})|(61x{10})#              && sip:$$@$n;user=phone       &&

/* DIGITMAP: Public international call, access code 6011 */

(6011x{7,15})|(6011x{7,15})#      && sip:$$@$n;user=phone       && t=8000
```

Figure 6: Sample dialing plan

Dialing function description

Dialing plan

As most IP Deskphone users are used to dialing digits to indicate the address of the destination, there is a need to specify the rule by which digits are transformed into a URI. The IP Deskphone with SIP Software dialing plan contains two sections delimited by two percent signs (%%).

```
+------------------------+---------------------------------------+
| declarations section   | user pre define variables and parameters |
| %%                     | section separator                     |
| digit maps             | list of digit maps                    |
+----------------------------------------------------------------+
```

Figure 7: Sample dialing plan declarations section

In the declaration section, the administrator can define the variables. The variables must start with a dollar ($) sign, followed by a number or a character, such as $1 or $a. There are two variables that are reserved by system. They are as follows:

$$: used for the collected digits if they match the pattern

$t : default timer

There must be a domain name defined and the domain name can be represented by any variable. In Figure 6: Sample dialing plan on page 73, the domain name is represented by $n.

The variable definitions take the form:

```
+--------------------------------------------+
| Name = value                               |
+--------------------------------------------+
```

Figure 8: Sample dialing plan variable definitions

For example:

$1="avaay.com"

$2="Avaya"

$3="."

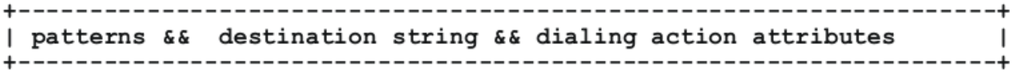
$4="com"

$5="Avaya.com"

$t=10000 (default timer is 10 seconds)

$a=Avaya.com

The second section of dialing plan contains the digit map. The digit map section has three subsections that are divided by a separator of two ampersands (&&).



```
+-------------------------------------------------------------------+
| patterns &&  destination string && dialing action attributes      |
+-------------------------------------------------------------------+
```

Figure 9: Sample dialing plan digit map section

The first part of a dialing plan contains a pattern defined with DRegex, which is used for matching the dialed number. The patterns are separated by the pipe (|) sign. The second part contains the result string used in the dial step. The third part defines the parameters used by UA in dialing action.

The following parameter is currently defined:

t=xxxx: After this timer expires, the number entered is automatically dialed. The timer starts after the first digit is entered and after it expires, the collected digits are automatically dialed out. xxxx is a decimal number in msec. The default timer is used when t is not specified in the digit map.

For example:

X{4} && sip:$$; phone-context=avaya.com;user=phone && t=7000

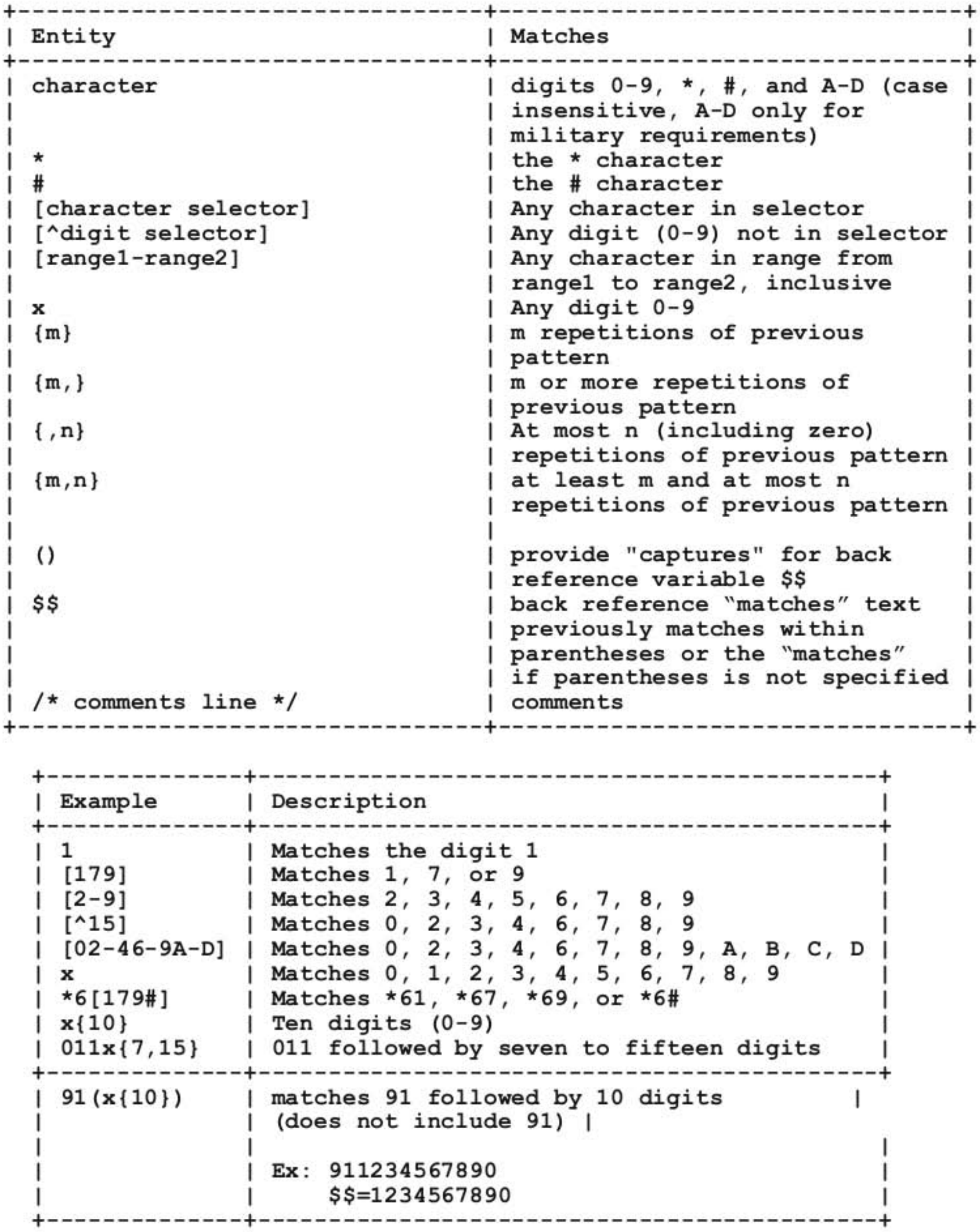
When the user presses any 4 digits, such as 4567, the following SIP URIs are generated because of the translation rule:

Sip:4567; phone-context=avaya.com;user=phone. The timeout of stopping the collection of digits is 7 seconds.

The pound sign (#) at the end of the digit map causes the IP Deskphone to dial the matched dialing plan immediately.

DRegex

The Digit Regular Expression (DRegex) syntax is a telephony-oriented mapping of Portable Operating System Interface (POSIX) Extended Regular Expressions (ERE). Users must take care not to confuse the DRegex syntax with POSI EREs, as they are not identical. In particular, there are many features of POSIX EREs that DRegex does not support. The dialing plan uses DRegex instead of ERE. The following rules demonstrate the use of DRegex.



```
+---------------------------------+-------------------------------+
| Entity                          | Matches                       |
+---------------------------------+-------------------------------+
| character                       | digits 0-9, *, #, and A-D (case |
|                                 | insensitive, A-D only for     |
|                                 | military requirements)        |
| *                               | the * character               |
| #                               | the # character               |
| [character selector]            | Any character in selector     |
| [^digit selector]               | Any digit (0-9) not in selector |
| [range1-range2]                 | Any character in range from   |
|                                 | range1 to range2, inclusive   |
| x                               | Any digit 0-9                 |
| {m}                             | m repetitions of previous     |
|                                 | pattern                       |
| {m,}                            | m or more repetitions of      |
|                                 | previous pattern              |
| {,n}                            | At most n (including zero)    |
|                                 | repetitions of previous pattern |
| {m,n}                           | at least m and at most n      |
|                                 | repetitions of previous pattern |
|                                 |                               |
| ()                              | provide "captures" for back   |
|                                 | reference variable $$         |
| $$                              | back reference “matches” text |
|                                 | previously matches within     |
|                                 | parentheses or the “matches”  |
|                                 | if parentheses is not specified |
| /* comments line */             | comments                      |
+---------------------------------+-------------------------------+

+--------------+--------------------------------------------+
| Example      | Description                                |
+--------------+--------------------------------------------+
| 1            | Matches the digit 1                        |
| [179]        | Matches 1, 7, or 9                         |
| [2-9]        | Matches 2, 3, 4, 5, 6, 7, 8, 9             |
| [^15]        | Matches 0, 2, 3, 4, 6, 7, 8, 9             |
| [02-46-9A-D] | Matches 0, 2, 3, 4, 6, 7, 8, 9, A, B, C, D |
| x            | Matches 0, 1, 2, 3, 4, 5, 6, 7, 8, 9       |
| *6[179#]     | Matches *61, *67, *69, or *6#              |
| x{10}        | Ten digits (0-9)                           |
| 011x{7,15}   | 011 followed by seven to fifteen digits    |
+--------------+--------------------------------------------+
| 91(x{10})    | matches 91 followed by 10 digits         |
|              | (does not include 91) |
|              |                                            |
|              | Ex: 911234567890                           |
|              |     $$=1234567890                          |
+--------------+--------------------------------------------+
```

Figure 10: DRegex rules

Downloadable WAV files

It is possible to customize the ring tones on the IP Deskphone with SIP Software. Up to five special ring tones can be downloaded from the provisioning server and stored on the IP Deskphone. The end user can select which ring tone they would like to implement.

In order to download these special files, the files must reside on the provisioning server and be specified in the SIP provisioning file. For more information, see Download the SIP Software to the provisioning server on page 32. The WAV files have a maximum size of 512 KB each for the IP Deskphone.

The file format is restricted to ITU-T A-law or u-law (8.0 kHz, 8-bit, mono or 16.0 kHz, 16 bit mono).

After the WAV files are downloaded to the IP Deskphone, the WAV file names appear in **Pref > Audio > Tones > Ring Pattern** (1 to 8 are standard ring tones, and 9 and above are WAV ring tones) and the WAV ring tones can then be selected to replace the standard ring tones.

For further information about downloadable WAV files, see the applicable IP Deskphone User Guide.

Configuration of the IP Deskphone using Web Manager

The administrator can access the SIP Software Web Manager through the web browser by typing the URL to start the SIP Software Web Manager configuration interface.

SIP Software for the IP Deskphone supports a basic text-based device config file mechanism through TFTP, FTP, and HTTP for device configuration. Although simple and efficient, the text-based config file is very limiting and challenging in larger enterprise or carrier (hosted) environments that have a large number of IP Deskphones and varying user requirements. Therefore, there is a significant need for enhanced tools and a centralized management entity that can remotely manage all the devices.

The SIP Software Web Manager adds flexible device configuration options by providing an easy to use web user interface for small to medium-size deployments, and provides an efficient solution for managing and provisioning the IP Deskphone.

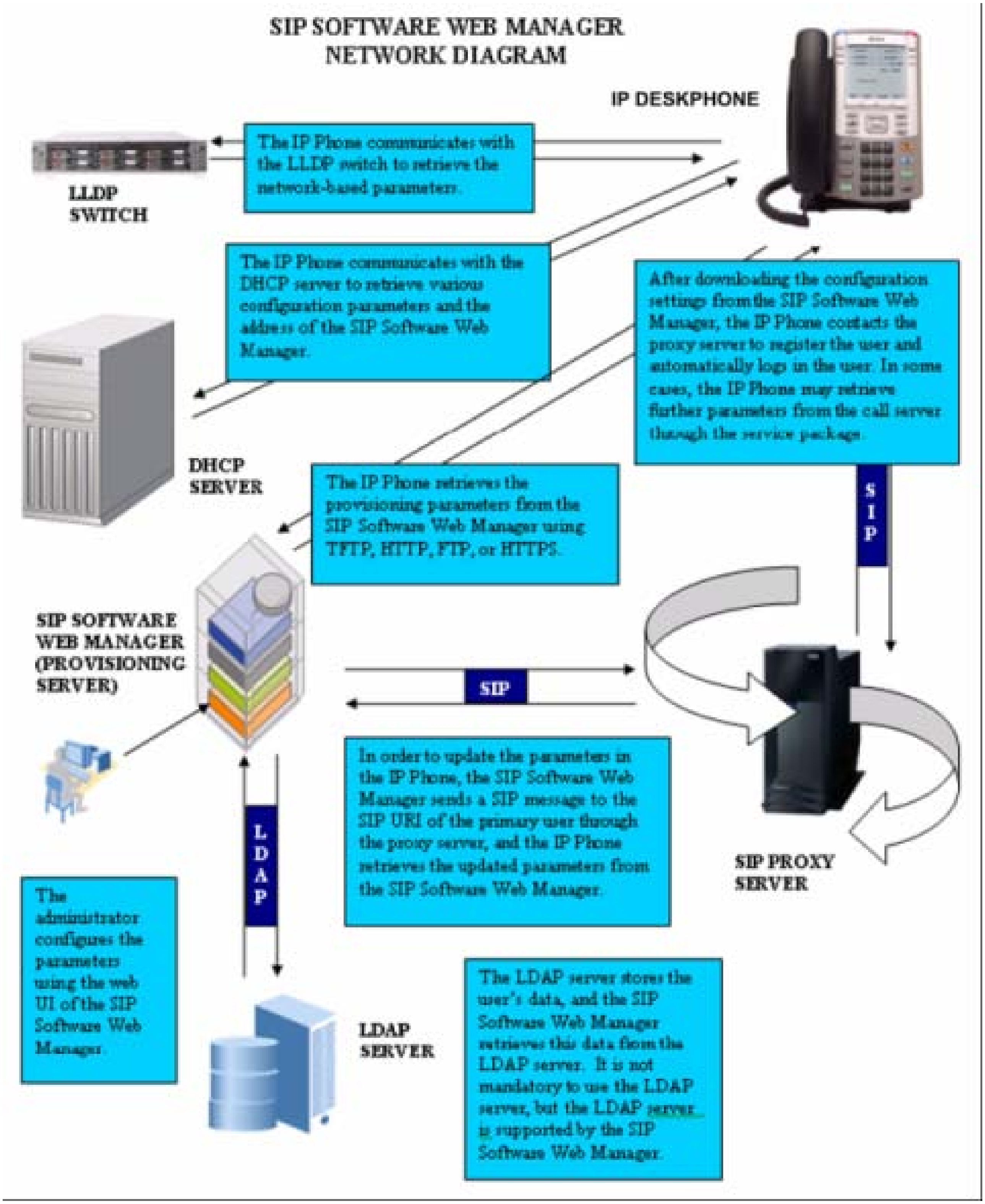
The four types of mechanisms that are supported to allows the IP Deskphone to access the configuration files are:

- TFTP
- FTP

- HTTP
- HTTPS

The SIP Software Web Manager provides and easy-to-use web interface for configuring the IP Deskphone and managing users, groups of users and domains. The SIP Software Web Manager can also generate configuration files, such as:

- 12xxSIP.cfg
- 11xxDeviceConfig.txt
- SIP<MAC>.txt
- dialplanx.txt
- <Users>.txt
- language files



The IP Phone communicates with the LLDP switch to retrieve the network-based parameters.

The IP Phone communicates with the DHCP server to retrieve various configuration parameters and the address of the SIP Software Web Manager.

After downloading the configuration settings from the SIP Software Web Manager, the IP Phone contacts the proxy server to register the user and automatically logs in the user. In some cases, the IP Phone may retrieve further parameters from the call server through the service package.

The IP Phone retrieves the provisioning parameters from the SIP Software Web Manager using TFTP, HTTP, FTP, or HTTPS.

In order to update the parameters in the IP Phone, the SIP Software Web Manager sends a SIP message to the SIP URI of the primary user through the proxy server, and the IP Phone retrieves the updated parameters from the SIP Software Web Manager.

The administrator configures the parameters using the web UI of the SIP Software Web Manager.

The LDAP server stores the user's data, and the SIP Software Web Manager retrieves this data from the LDAP server. It is not mandatory to use the LDAP server, but the LDAP server is supported by the SIP Software Web Manager.

Figure 11: SIP Software Web Manager network diagram

The preceding figure illustrates the different components, in the network, that communicate with the SIP Software Web Manager. When the IP Deskphone restarts, the IP Deskphone retrieves the network-based parameters (VLAN ID, Auto negotiation parameters) from the LLDP switch (if the LLDP switch is enabled). The IP Deskphone communicates with the DHCP server to retrieve the various configuration parameters and the URL of the SIP Software Web Manager. The DHCP server sends the IP address and the network configuration parameters to the IP Deskphone, and then contacts the SIP Software Web Manager to download the configuration files. If the feature is enabled through the SIP<MAC>.cfg or 12xxDeviceConfig.txt files, automatic login occurs. The IP Deskphone registers with the proxy server, downloads the service package, and then the user is logged in automatically. If multiple users are configured in the SIP Software Web Manager, then multiple SIP user accounts are registered at the same time.

The precedence rule for the configuration is:

1. Manual
2. LLDP
3. Service Package (Received from the SIP Call Server)
4. DHCP
5. SIP Software Web Manager (TFTP, FTP, HTTP, or HTTPS)

After the configuration phase, the configuration parameters can be changed manually on the IP Deskphone. All the manual configuration parameters remain unchanged until they are changed to an autoprovisioning option.

The following files are downloaded by the IP Deskphone:

- 12xxSIP.cfg
- 11xxDeviceConfig.txt
- Domains.txt
- DialPlan.txt
- language files
- software files
- SIP<MAC>.txt
- <User>.txt

The SIP Software Web Manager generates new configuration files. The 12xxSIP.cfg file is the first file provisioned on the IP Deskphone and contains references to the following files:

- 12xxDeviceConfig.txt
- Domains.txt
- software

- dial plan
- languages

The USER CONFIG section invokes the MAC-specific file which contains IP Deskphone-specific information, mapping between lines, and user-specific files. The user-specific files are downloaded for each user with the user-specific parameters.

For more information on the SIP Software Web Manager, see *Using the SIP Software Web Manager, NN43112-500*.

Chapter 7: Configure the DHCP Server

The Avaya 1200 Series IP Deskphones support two basic Dynamic Host Configuration Protocol (DHCP) mechanisms to provide configuration information to the IP Deskphones. These mechanisms are the following:

- Normal DHCP
- DHCP VLAN phase

Normal DHCP

The normal DHCP is used to configure standard IP parameters such as IP address, NetMask, default gateway, and DHCP lease parameters. The message sequence consists of Discover, Offer, Request, and Acknowledge. The IP Deskphones can also insert an optional phase. To include an optional phase, the first phase is used to discover and configure the voice VLAN using a Avaya proprietary method. The second phase then proceeds normally on the discovered VLAN. If the DHCP VLAN discovery is not used, then there is only a single phase.

IP Deskphone DHCP VLAN Phase

The DHCP site and vendor specific options contain VLAN information to configure VLANs. The VLAN parameters are text string embedded in the standard DHCP Vendor and Site Specific options. You can acquire the VLAN parameters using 802.1ab and IP address parameters using DHCP.

If the IP Deskphone does not contain VLAN configuration provisioned manually or through LLDP, the IP Deskphone attempts to determine the VLAN during DHCP VLAN Phase. If the IP Deskphone does find a VLAN configuration it proceeds to the DHCP Configuration Phase. If the VLAN Phase (VLAN configured through DHCP) is successful, then the VLAN Phase finishes with a final DHCP. Also, a release message appears after the completion of the Configuration Phase.

Following is the procedure to configure the Voice VLAN using DHCP assuming VLAN is not configured using any other method:

1. The IP Deskphone sends a DHCP request using an untagged (no VLAN) packet during any of the following scenarios:
    - The customer network is configured to handle untagged packets; for example, retag them to a specific VLAN.

- The DHCP request contains standard IP Deskphone IP DHCP option requests from the point when the IP Deskphone does not receive the VLAN information. These options include the Vendor Specific and all Site Specific options.

2. The DHCP server receives the request. If the server is configured, the DHCP server returns a DHCP Offer message with a special text string in the Vendor Specific option or one of the Site Specific options.

   Following is the format of the text in the option:

   VLAN-A:XXX+YYY+ZZZ+…

   where VLAN-A is a substring followed with VLAN information. XXX, YYY, ZZZ are the numbers of the supported VLANs. They can be from 1 to 10 different VLANs; each VLAN is separated with the symbol +.
3. After receiving the DHCP Offer message, the IP Deskphone scans each Vendor and Site Specific options for the string VLAN-A.
4. If the IP Deskphone finds the VLAN-A string, it tries each VLAN, and in turn XXX, YYY, ZZZ searches for a DHCP server.
5. The search is done by sending a DHCP Discover message looking for a DHCP Offer message as a response.
6. If the IP Deskphone finds a response, it discontinues its DHCP exchange on the untagged channel and continues its DHCP exchange on the “discovered” VLAN.
7. If there is no response, the initial untagged Discover message assumes there is no VLAN configuration information available from the DHCP server and continues using untagged packets. When the IP Deskphone sends its first DHCP Discover message, it does not know if it can find VLAN configuration information. If it does discover VLAN information, it continues the VLAN configuration as described above. If it does not find any VLAN information, then it assumes there is only a Configuration Phase.

DHCP options

The DHCP protocol provides option mechanisms for the client and server to exchange information in addition to the standard Bootstrap Protocol (BOOTP) information. This section describes the client and server options supported by the IP Deskphone.

- IP Deskphone to Server options on page 85
- Server to IP Deskphone options on page 86

IP Deskphone to Server options

When a DHCP client sends a DHCP Discover and Request message, it includes a list of options as part of the request. The IP Deskphone DHCP client sends the following options:

Option	Description
12	Specifies the Hostname. By default, the Hostname is “T”+MAC Address; for example, T001765FDBF1D. The Hostname can be manually provisioned using the keypad.
53	Specifies the DHCP Message Type.
55	Specifies the messages to tell the server which options the IP Deskphone is requesting. It appears in the Discover and Request. The SIP software requests the following options • 1 - IPv4 Subnet Mask • 3 - Router • 6 - Domain Name Server • 15 - Domain Name • 28 - Broadcast Address • 43 - Vendor Specific Information • 58 - Renewal Time • 59 - Rebinding Time • 66 - TFTP Server Name. The client treats this more generically as a request for the provisioning server name and protocol. • 99 - Must not be included • 128, 131, 144, 157, 188, 191, 205, 219, 223 — Specifies old site specific options. Recovered by IANA according to RFC 3942 and must not be used for new installations. • 224, 227, 230, 232, 235, 238, 241, 244, 247, 249, 251, 254 — Specifies site specific options.
57	Specifies maximum DHCP message size. The maximum message size is 1190 bytes.
60	Sends “Avaya-SIP-Phone-A” as the Vendor Identifier.
61	Specifies Client Identifier (MAC Address).

Server to IP Deskphone options

The DHCP server can send any option to the IP Deskphone as part of the DHCP Offer message. The IP Deskphone accepts the following options:

	DHCP Option	Description
IPv4 Address		
Net mask	1	
Router Option	3	
Domain Name Server	6	Accepts the first two DNS addresses.
Domain Name	15	
Broadcast Address	28	
Vendor Specific Option	43	
DHCP Renewal Time	58	
DHCP Rebinding Time	59	
TFTP Server Name	66	Two forms of the server name are supported. If a dotted-decimal IP address is returned, it is assumed to point to a TFTP server. A full URL can also be provided to specify a protocol and FQDN.
Old Site Specific Options	128, 131, 144, 157, 188, 191, 205, 219, 223	Options are supported, but not recommended, for new installations. These options are reclaimed according to RFC 3942.
Site Specific Options	128, 131, 144, 157, 188, 191, 205, 219, 223	

Multiple DHCP Servers

It is possible that two or more DHCP servers can respond to the DHCP Discover message. When the IP Deskphone sends a Discover message, it waits for 1 second to collect all the responses. If there is more than one response, the IP Deskphone selects the response with the longest lease time. If the lease time is identical, the first response is selected.

Chapter 8: Install the IP Deskphone

Complete instructions to install the Avaya 1200 Series IP Deskphone, including detailed figures and applicable warnings, are given in the Avaya IP Deskphones User Guides.

The steps for installing the Avaya 1200 Series IP Deskphone are summarized in the following procedure.

Installing the IP Deskphone

1. Remove the stand cover. Pull upward on the center catch and remove the stand cover. The cable routing tracks are now accessible.
2. Connect the AC power adapter (optional). Connect the adapter to the AC adapter jack in the bottom of the IP Deskphone. Form a small bend in the cable, and then thread the adapter cord through the channels in the stand.
3. Install the handset. Connect the end of the handset cable with the short straight section into the handset. Connect the end of the handset cable with the long straight section to the back of the IP Deskphone, using the RJ-9 handset jack. Form a small bend in the cable, and then thread the handset cord through the channels in the stand so that it exits behind the handset on the right side, in the handset cord exit in the stand base.
4. Install the headset (optional). If installing a headset, plug the connector into the RJ-9 headset jack on the back of the IP Deskphone, and thread the headset cord along with the handset cord through the channels in the stand, so that the headset cord exits the channel.
5. Install the Ethernet cable. Connect one end of the supplied Ethernet cable to the back of the IP Deskphone using the RJ-45 connector and thread the network cable through the channel.
6. Install the Ethernet cable connecting the PC to the IP Deskphone (optional). If connecting PC Ethernet through the IP Deskphone, connect one end of the PC Ethernet cable to the IP Deskphone using the RJ-45 connector and thread it through the channel. Connect the other end to the LAN connector on the back of the PC.
7. Install additional cables. Connect the Ethernet cable to the LAN Ethernet connection. If using an AC power adapter, plug the adapter into an AC outlet.
8. Wall-mount the IP Deskphone (optional). The IP Deskphone can be mounted either by: (method A) using the mounting holes on the bottom of the IP Deskphone stand, or (method B) using a traditional-style wall-mount box with RJ-45 connector and 15-cm (6-inch) RJ-45 cord (not provided).
9. Replace the stand cover. Ensure that all cables are neatly routed and press the stand cover into place until a click is heard.
10. Put the IP Deskphone in the wall-mount position (optional). If the IP Deskphone is to be mounted on the wall, put it in the wall-mount position by holding the tilt lever and pressing the IP Deskphone towards the base until the IP Deskphone is parallel with the base. Release the tilt lever and continue to push the IP Deskphone towards the base until an audible click is heard. Ensure the IP Deskphone is securely locked in position.

The following figure shows the connections on the IP Deskphone.

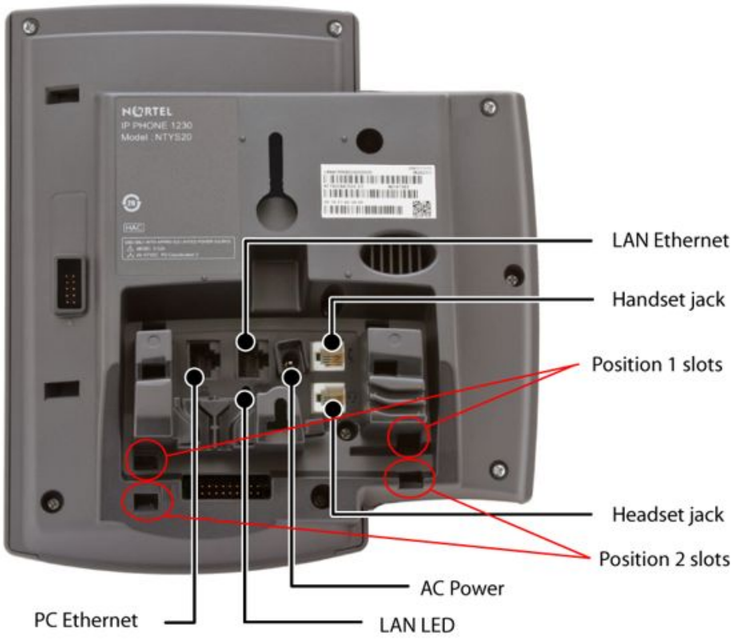


Figure 12: IP Deskphone connections

Chapter 9: Upgrade and convert the IP Deskphone software

Introduction

This chapter describes how to upgrade an Avaya 1200 Series IP Deskphone with UNIStim software to SIP Software.

In order to upgrade an IP Deskphone with UNIStim software, first determine if you have the minimum UNIStim software release on the IP Deskphone (062AC5L). If your IP Deskphone is installed with the minimum version of UNIStim software, proceed to the section Convert UNIStim software to SIP Software on the IP Deskphone on page 95. If your IP Deskphone is not installed with the minimum version of UNIStim Software, proceed to the section Upgrade to the minimum UNIStim Software on page 90.

To convert the firmware on the IP Deskphone from SIP to UNIStim, see the section Maintenance on page 279.

Upgrade the SIP Software on the IP Deskphone

Use the following procedures to upgrade existing SIP Software to new SIP Software on the IP Deskphone.

Download the SIP Software to the provisioning server

To download the SIP Software, perform the following procedure.

Downloading SIP Software for the IP Deskphone from the Avaya Web site

1. Go to http://www.avaya.com/support.
2. Log on to the Avaya Web site with a valid Avaya User ID and Password.

   The **Avaya Support** page appears.
3. Enter IP Deskphone type in the **Knowledge and Solution Engine** box.
4. Select **Software** in the **All types** scroll-down menu.

5. Press the gray arrow at the end of the **Knowledge and Solution Engine** box to obtain the **Search Results**.
6. From the **Search Results**, select the appropriate version of the SIP Software for the IP Deskphone; for example, **SIP IP Phone 11230 Release SIP12x004.01.03.00.bin**.
7. Place the selected software on the provisioning server.

Modify the SIP provisioning file

Use the following procedure to modify the SIP provisioning file, which exists on the provisioning server.

Modifying the SIP provisioning file

1. Under the firmware [FW] section of the SIP Provisioning file, increase the VERSION number (for example 06A5C39d26).
2. Under the firmware [FW] section of the SIP Provisioning file, modify the FILENAME of the new file you want to upload to the IP Deskphone.

   Important:

   The VERSION number must be the same as the FILENAME (do not include the .bin extension).

   For example, if the FILENAME is SIP1140e03.00.33.04.bin, then the VERSION must be SIP1140e03.00.33.04.
3. Invoke the upgrade mechanism.

   Use one of the next three methods to invoke a software upgrade on the IP Deskphone with SIP Software.

   a. Power off and power on the IP Deskphone.

   b. Select **Services, System, Check For Updates** on the IP Deskphone.

   c. Allow for an automatic check for updates to occur. (See AUTO_UPDATE under Feature configuration commands on page 43).

   Any of these actions causes the IP Deskphone to contact the provisioning server and attempt to read the Provisioning file. A Soft Reset (**Srvcs > System > Reset Phone**) does not cause the IP Deskphone to retrieve the Provisioning file and therefore does not cause a software upgrade.

Upgrade to the minimum UNIStim Software

The Avaya 1200 Series IP Deskphones can be ordered with UNIStim software installed or with SIP Software installed. You can convert the software on an IP Deskphone from UNIStim to

SIP. To successfully convert the software from UNIStim to SIP, the UNIStim software version on your IP Deskphone must be 062AC5L or higher.

Identify the current version of UNIStim software

For a new Avaya 1200 Series IP Deskphone, use the following procedure to check the current version of UNIStim software to determine the version number of the UNIStim software on the IP Deskphone.

For an IP Deskphone already in use, follow Checking the UNIStim software version on a new IP Deskphone on page 91 to determine the version number of UNIStim software on an IP Deskphone.

Checking the UNIStim software version on a new IP Deskphone

1. After assembling the IP Deskphone and turning it on, the display on the IP Deskphone goes through the following sequence:

    - Avaya splash screen appears
    - Avaya sonic sound plays
    - Avaya banner appears

    Following the Avaya banner, the software version appears in the display (**F/W version**).

2. Note the UNIStim software version number and write it down. Compare the version number to the minimum-required UNIStim software version (062AC5L).

    If the version number is equal to or higher than 062AC5L, see Convert UNIStim software to SIP Software on the IP Deskphone on page 95.

    If the number is lower than 062AC5L, go to the section Upgrade UNIStim software to the minimum required UNIStim software on page 92 and follow the instructions to upgrade an IP Deskphone to the minimum-required version of UNIStim software before a conversion to SIP Software.

Checking the UNIStim software version on an IP Deskphone already in use

1. Press the **Globe/Services** key on the IP Deskphone twice quickly.

    If the admin password prompt appears, enter the password **26567*738**.

    The **Local Tools** menu appears:

    Table 8: Local Tools menu

1. Preferences
2. Local Diagnostics
3. Network Configuration

4. Lock Menu

To make a selection, press the number associated with the menu item, or use the **Navigation key cluster**

to scroll through the menu items. Press the **Select** key to select the highlighted menu item.

Table 9: Using the Navigation key cluster to navigate in the Local Tools menu

Key	Action
Down	Moves highlight down
Up	Moves highlight up
Right	Selected current menu item
Left	Closes menu
Select key (center of cluster)	Selects current menu item

To close this menu, use the **Quit** key.

2. Select **2. Local Diagnostics** in the Local Tools menu by pressing the **Select** key in the Navigation key cluster or by pressing the number **2**.
3. Select **IP Set and DHCP Information** by pressing the **Select** key in the Navigation key cluster or by pressing the number **2**.
4. Use the down arrow in the Navigation key cluster to scroll down the menu to **Software Version**.
5. Note the UNIStim software version number and write it down.

   Compare the version number to the minimum-required UNIStim software version (062AC5L).

If the version number is equal to or higher than 062AC5L, go to the section Convert UNIStim software to SIP Software on the IP Deskphone on page 95.

If the number is lower than 062AC5L, see Upgrade UNIStim software to the minimum required UNIStim software on page 92 and follow the instructions to upgrade an IP Deskphone to the minimum-required version of UNIStim software before you convert to SIP Software.

Upgrade UNIStim software to the minimum required UNIStim software

Use either of the following two methods to upgrade UNIStim software.

1. UFTP download initiated by the server if the server supports this method of upgrading UNIStim software. Refer to the appropriate documentation for your Call Server for instructions on using this method.
2. TFTP download on bootup.

If necessary, use the following procedure to configure the TFTP server.

Configuring the TFTP server

1. The 1200 Series IP Deskphones always execute the TFTP download at bootup if a TFTP IP address is configured on the IP Deskphone after being initiated by the telephony Call Server.
2. Go to the TFTP server and create the 12xx.cfg provisioning file. The 12xx.cfg provisioning file is a clear text file. Create the provisioning file as shown in the next table.

   Table 10: Sample 12xx.cfg provisioning file

   ```
   [FW]
   DOWNLOAD_MODE FORCED
   VERSION 062AC5L
   FILENAME SIP12x004.01.03.00.bin
   ```

   This configuration file forces the software download of SIP12x004.01.03.00.bin.
3. Download and copy the software to the TFTP server directory.

   To download the UNIStim software for the IP Deskphone from the Avaya Web site:

   a. Go to http://www.avaya.com/support.

   b. Log on to the Avaya Web site with a valid Avaya User ID and Password.

      The **Support** page appears.

   c. Enter the IP Deskphone type in the **Knowledge and Solution Engine** box.

   d. Select **SOFTWARE** in the **ALL TYPES** scroll-down menu.

   e. Press the gray arrow at the end of the **Knowledge and Solution Engine** box to obtain the **Search Results**.

   f. From the **Search Results**, select the appropriate version of the UNIStim software for the IP Deskphone, for example, **IP Phone 1230 Release 0625C23**.

   g. Place the selected software in the correct directory on the provisioning server.
4. In the IP Deskphone **Network Configuration** menu, change the **TFTP server address** and enter the correct TFTP server address.

This can be the provisioning server as defined in the chapter Configure the provisioning server on page 31.

5. Select the **Apply&Reset** soft key to save the configurations and reset the IP Deskphone.

   The IP Deskphone downloads the software file. The display shows **[FW] reading…**

   If the download is successful, the display shows **[FW] writing…** and the blue LED flashes.

   After the software image is downloaded to the IP Deskphone, the display shows **[FW] finished...**, the blue LED stops flashing, and the IP Deskphone resets.

   The IP Deskphone registers to the TPS with the new software version.

   If the upgrade is unsuccessful, see the chapter Diagnostics and troubleshooting on page 283 in the section **Download failures**.

Follow the next procedure to download the minimum required version of UNIStim software automatically through TFTP on bootup.

Downloading UNIStim software automatically through TFTP on bootup

1. Press the **Globe/Services** key on the IP Deskphone twice quickly.

   If the admin password prompt appears, enter the password **26567*738**

   The **Local Tools** menu appears.

   Table 11: Local Tools menu

1. Preferences
2. Local Diagnostics
3. Device Settings
4. Lock Menu

2. Select **3. Device Settings** from the **Local Tools** menu.

   The **Device Settings** screen appears.

3. If you are using DHCP, select **Yes**.

   If you are manually configuring the IP address, netmask, and gateway address, select **No**.

4. If the DHCP option is configured, the IP address is automatically obtained.
5. Configure the TFTP IP address within the IP Deskphone Device Settings menu.

   This can be the provisioning server as defined in the chapter Configure the provisioning server on page 31.

6. Select the **Apply&Reset** soft key to save the settings and reset the IP Deskphone.

The IP Deskphone downloads the software file. The display shows **[FW] reading…**

If the download is successful, the display shows **[FW] writing…** and the blue LED flashes.

After the software image is downloaded to the IP Deskphone, the display shows **[FW] finished...**, the blue LED stops flashing, and the IP Deskphone resets.

If the upgrade is unsuccessful, see the chapter Maintenance on page 279 in the section **Download failures**.

Convert UNIStim software to SIP Software on the IP Deskphone

The Avaya 1200 Series IP Deskphones can be ordered with UNIStim software installed or with SIP Software installed. If an IP Deskphone is installed with UNIStim software, it runs with SIP Software only if the software is converted from UNIStim to SIP. If the procedure to determine the UNIStim version number is completed, and, if necessary, the procedure to upgrade the UNIStim software is completed, an IP Deskphone can be converted from UNIStim software to SIP Software.

Compare the version number to the minimum required UNIStim software version (062AC5L).

The conversion must be performed using TFTP.

⚠ **Warning:**

The TFTP download and upgrade of the Flash memory on the IP Deskphone can take a significant amount of time (possibly up to 10 minutes). Do not unplug or reboot the IP Deskphone during the process.

The following procedure explains how to download the SIP Software from the Avaya Web site.

Downloading SIP Software for the IP Deskphone from the Avaya Web site

1. Go to http://www.avaya.com/support.
2. Log on to the Avaya Web site with a valid Avaya User ID and Password.

   The **Support** page appears.
3. Enter the IP Deskphone type in the **Knowledge and Solution Engine** box.
4. Select **SOFTWARE** in the **ALL TYPES** scroll-down menu.
5. Press the gray arrow at the end of the **Knowledge and Solution Engine** box to obtain the **Search Results**.

6. From the **Search Results**, select the appropriate version of the SIP Software for the IP Deskphone for example, **IP Phone 1230 Release SIP12x004.01.03.00.bin**.
7. Place the selected software on the provisioning server.

Perform the following procedure to convert the UNIStim software to SIP Software on the IP Deskphone.

Converting UNIStim software to SIP Software using TFTP

1. Run the TFTP server (for example Tftpd32.exe).
2. Place software and configuration files in the folder of the TFTP server (for example 12xx.img F/W file and 12xx.cfg file) that contains the following lines:

   Table 12: Sample 12xx.cfg configuration file

   ```
   [FW]
   DOWNLOAD_MODE AUTO
   VERSION SIP12x004.01.03.00.bin
   FILENAME 12xx.img
   ```

3. Configure the IP Deskphone Device Settings TFTP IP address to the IP address where your TFTP server is running.

   After you are finished the configuration, the IP Deskphone reboots and sends a request to the TFTP server.

4. Select the **Apply&Reset** soft key to save the settings and reset the IP Deskphone.

   The following messages display on the IP Deskphone as the IP Deskphone cycles through the conversion process, one after the other:

   a. [FW] Reading...
   b. [FW] Writing...
   c. [FW] Finished...

   The IP Deskphone then boots up with SIP Software.

   If the conversion is unsuccessful, see the chapter Maintenance on page 279.

1. TFTP file transfer takes approximately 15 seconds.
2. File writing takes 2.5 minutes. The IP Deskphone displays the message **[FW] writing…** and the blue **Data Waiting** LED flashes.
3. After the new SIP Software writing is finished, the blue LED stops flashing and the IP Deskphone displays **[FW] finished** and then reboots.
4. The first time the SIP Software boots, the SIP Software performs a Flash File System conversion that takes 2.5 minutes.

Chapter 10: Voice Quality Monitoring

Feature overview

Proactive Voice Quality Monitoring (PVQMon or VQMon) allows the Avaya 1200 Series IP Deskphone with SIP Software to report voice quality statistics to a server in the network. The IP Deskphone with SIP Software collects various voice quality statistics, for example, packet loss, and sends the voice quality statistics to the server at regular intervals during a call. A subset of these statistics is also available for the user to view on the IP Deskphone by selecting the Audio softkey and then the Monitor Audio Quality menu item.

VQMon set-up

Configure the following parameters on the IP Deskphone with SIP Software to connect to the server and send the PVQMon statistics.

1. Enable the feature. To enable the feature, configure the VQMON_PUBLISH parameter in the device configuration file (see VQMon configuration commands on page 66).
2. Configure the IP address of the PVQMon server. Configure the IP address of the PVQMon server in either of the following settings:
    a. Configure VQMON_PUBLISH_IP through the device configuration file (see VQMon configuration commands on page 66).
    b. Configure PVQMon IP in Device Settings (see Table 47: PVQMon IP configuration on page 125)
3. Configure the remainder of the VQMon parameters in the device configuration file (see VQMon configuration commands on page 66). These parameters provide threshold information to the IP Deskphone with SIP Software. A report is sent to the server when these thresholds are exceeded.

Server set-up

The IP Deskphone with SIP Software works with Telchemy server software. The name of the software is SQmediator and is available through Telchemy (http://www.telchemy.com). The minimum version required is release 1.0.

How VQMon works

The IP Deskphone with SIP Software gathers statistics about the current call when VQMon is enabled. Statistics are also gathered regarding the quality metrics of the current call. The call-related statistics contain condensed information about the SIP Session Description Protocol (SDP), the Call ID, the local and remote address, voice quality-related statistics, Zulu times for start-time and the time the report was sent.

The voice quality-related statistics include jitter, packet loss, delay, burst gap loss, listening R-factor, R-LQ, R-CQ, MOS-LQ and MOS-CQ. See Table 13: Glossary of RTCP XR metrics on page 98. More information on each of these metrics is provided in RFC3611 “RTP Control Protocol Extended Reports (RTCP XR)”.

When the IP Deskphone detects that a particular voice quality metric has exceeded a threshold (defined in the device configuration file), the IP Deskphone sends a message to the server indicating that there is an issue. If the issue persists then the IP Deskphone reports another message indicating that there is an exceeded value at regular intervals. This happens continuously until the voice quality metric falls below the threshold value. As well, the IP Deskphone can send regular reports of the voice quality at time intervals defined in the device configuration file.

Table 13: Glossary of RTCP XR metrics

Metric	Description
Burst	A period of high packet losses and / or discards. A burst is calculated in milliseconds.
Conversational R-factor	Voice quality metric based on burst packet loss and vocoder selection.
Delay	One way delay which includes end-to-end delay, jitter buffer delay and packetization delay. Delay is calculated in milliseconds.
Inter-arrival jitter	The variation in packet arrival times due to transmission (routing, queuing delay) through the network. Jitter is calculated in milliseconds.

Metric	Description
Listening R-factor	Voice quality metric based on burst packet loss, transmission delay and burst loss.
MIU	Media Information Unit. MIU is a concept from VQMon. An MIU can be any size down to a 10 millisecond (8 sample) block. An MIU means a frame in the i200x implementation.
MOS	Mean Opinion Score. A subjective measurement of the voice quality of a voice call.
MOS_CQ	The VQMon conversational quality MOS score calculated for a call channel.
MOS_LQ	The VQMon listening quality MOS score calculated for a call channel.
Packet loss rate	The percentage of total packets loss versus packets received.
R-factor	A measurement of voice quality based on network impairments including burst packet loss, delay and encoding/decoding algorithm selection.

End of call report

The IP Deskphone with SIP Software sends a report using VQMON Publish message to the proxy. The proxy redirects the publish ID described within the report. An end-of-call report is always generated if VQMON is enabled. IP Deskphones with SIP Software do not negotiate or exchange messages with the device defined using PUBLISH_IP options.

Session interval report

The IP Deskphone with SIP Software can send voice quality reports at time intervals defined in the device configuration file. The minimum and default time interval is 60 seconds. If the IP Deskphones with SIP Software send session interval reports more frequently, then a threshold violation has occurred.

Alert interval report

When an IP Deskphone with SIP Software detects that a voice quality metric has exceeded a threshold, the IP Deskphone with SIP Software initiates a timer which sends a message to the server every 5 seconds. When all voice quality metrics fall below the threshold values, the IP Deskphone with SIP Software stops sending VQMON Publish messages with the report. The alert interval report does not differ from the session interval reports or end-of-call reports.

Chapter 11: Device Settings on the IP Deskphone with SIP Software

Important:

An Avaya 1200 Series IP Deskphone with SIP Software displays different menus than an IP Deskphone with UNIStim software.

Introduction

This chapter describes how to configure the **Device Settings** parameters on the IP Deskphone with SIP Software. This includes items such as the IP address, the subnet mask, and the gateway IP address of the IP Deskphone.

The **Device Settings** parameters are listed below in the order they appear on the Device Settings menu of the IP Deskphone with SIP Software. Read the section at the end of this chapter that explains how to provision the Device Settings parameters. If you are familiar with the Device Settings parameters, skip the last section in the chapter and proceed to the provisioning instructions.

- Enable 802.1x (EAP)
- Device ID
- Password
- Enable 802.1ab (LLDP)
- DHCP: Yes, No
- SET IP
- Net Mask
- Gateway
- DNS IP1
- DNS IP2
- Ntwk Port Speed: Auto, 10BT, 100BT
- Ntwk Port Duplex: Auto, Force Full, Force Half
- Disable Voice 802.1Q

- Voice VLAN: No VLAN, Manual
- VLAN Filter
- Ctrl Priority bits: Auto, 0–7
- Media Priority bits: Auto, 0–7
- Disable PC Port
- PC Port Speed: Auto, 10BT, 100BT
- PC Port Duplex: Auto, Force Full, Force Half
- Disable Data 802.1Q
- Data VLAN: No VLAN, Manual (value from 1 to 4094)
- Data Priority bits: Auto, 0–7
- PC-Port Untag al
- Cached IP
- Ignore GARP
- Provisioning: Server URL, Protocol (TFTP/FTP/HTTP), Device ID, Password
- PVQMon IP
- NAT Traversal: NAT Signal (None/ SIP Ping/ STUN), NAT Media (None/ STUN), NAT TTL, STUN S1 IP, STUN S2 IP
- SSH: Yes, No
- SFTP: Yes, No

802.1x (EAP) Port-based network access control

Extensible Authentication Protocol (EAP) supports multiple authentication methods and represents a technology framework that facilitates the adoption of Authentication, Authorization, and Accounting (AAA) schemes, such as Remote Authentication Dial In User Service (RADIUS). RADIUS is defined in RFC2865. The IP Deskphone with SIP Software supports only the MD5 authentication method.

802.1x defines the following three roles:

1. Supplicant—an IP Deskphone that requires access to the network to use network services.
2. Authenticator—the network entry point to which the supplicant physically connects (typically a Layer 2/3 switch). The authenticator acts as the proxy between the supplicant and the authentication server. The authenticator controls access to the network based on the authentication status of the supplicant.
3. Authentication server—performs authentication of the supplicant.

Enable and disable Network-level authentication through the EAP configuration menu.

The RADIUS server is the authentication server and performs the actual authentication of the supplicant. The following EAP methods are supported:

- EAP-MDS on page 224
- EAP-TLS on page 224
- EAP-PEAP on page 224

The following options are available for the administrator:

- When EAP-MD5 is selected, the administrator is prompted to enter ID1 and Password
- When EAP-PEAP is selected, the administrator is prompted to enter ID1, ID2, and Password. If the administrator enters only ID1, then ID2 contains same value of ID1.
- When EAP-TLS is selected, the administrator is prompted to enter ID1. If SCEP is used to install the device certificate, the administrator is required to enter CA Server (URL of the SCEP service), the Domain Name which the IP Deskphone belongs to and optionally the Hostname.
- When Disabled mode is selected, the existing IDs and Passwords are erased.

Authorization

If 802.1x is configured and the IP Deskphone is physically connected to the network, the IP Deskphone (supplicant) initiates 802.1x authentication by contacting the Layer 2/3 switch (authenticator). The IP Deskphone also initiates 802.1x authentication after the Ethernet connection (network interface only) is restored following a network link failure.

However, if the IP Deskphone resets, it assumes the Layer 2 link has remained in service and is authenticated.

The IP Deskphone fails to authorize if the DeviceID and the IP Deskphone passwords do not match the DeviceID and IP Deskphone passwords provisioned on the RADIUS Server. The Layer 2 switch (authenticator) locks out the IP Deskphone and network access is denied. If this happens during reauthorization, all phone services are lost. The connected PC operates as normal.

Device ID

The Device ID is for use with the 802.1x (EAP) protocol. If the 802.1x (EAP) is not used, then there is no prompt to enter the Device ID.

Password

The Password is for use with the 802.1x (EAP) protocol. If the 802.1x (EAP) is not used, there is no prompt to enter the Password.

802.1ab Link Layer Discovery Protocol

802.1ab Link Layer Discovery Protocol (LLDP) is a standard for discovering the physical topology between neighboring devices. 802.1ab LLDP defines a standard method for Ethernet network devices, such as switches, routers, and IP Deskphones to advertise information about themselves to other nodes on the network and to store the information they discover in a Management Information Base (MIB).

802.1ab (LLDP) takes advantage of the VLAN Name and Network Policy TLVs, and provides an automatic configuration of the IP Deskphone network policy parameters. Key parameters, such as VLAN ID, L2 priority, and DSCP values are received from the switch and are automatically configured in the IP Deskphone.

802.1ab Link Layer Discovery Protocol (LLDP) provides the following functionality

- Periodic transmission of advertisements containing device information, device capabilities and media specific configuration information to neighbors attached to the same network.
- Reception of LLDP advertisements from its neighbors.
- Implementation of behavioral requirements specified by Link Layer Discovery Protocol Media Endpoint Discovery (LLDP-MED).
- Storage of received data in local data structures, for example, in MIB modules.

TLVs

The information fields in each MIB are contained in a Link Layer Discovery Protocol Data Unit (LLDPDU) as a sequence of short, variable-length, information elements known as TLVs that each include type, length, and value fields. Each LLDPDU includes several mandatory TLVs plus optional TLVs. Optional TLVs may be inserted in any order.

The IP Deskphone supports both the transmit and receive LLDP mode.

Transmit direction:

An LLDPDU transmitted by the IP Deskphone supports the following TLVs:

1. Chassis ID
2. Port ID
3. Time To Live
4. End of LLDPPDU
5. Port Description
6. System Description
7. System Capabilities
8. Port VLAN ID
9. Port And Protocol VLAN ID
10. VLAN Name
11. Protocol Identity
12. MAC/PHY Configuration Status
13. Power Via MDI
14. Link Aggregation
15. Maximum Frame Size
16. LLDP-MED Capabilities
17. Network Policy
18. Extended Power-via MDI
19. Inventory Software Revision
20. Inventory Manufacturer Name
21. Inventory Model Name

Receive direction:

The IP Deskphone expects to receive the following TLVs:

1. Chassis ID
2. Port ID
3. Time To Live
4. End of LLDPPDU
5. System Capabilities
6. VLAN Name
7. MAC/PHY Configuration Status
8. LLDP-MED Capabilities

9. Network Policy
10. Location Identification

Table 14: TLV formats

TLV	Fields
Chassis ID	Length = 6 Chassis Subtype = 5 [IP Address] Chassis ID = IP Deskphone IP Address
Port ID	Length = 7 Port Subtype = 3 [MAC Address] Port ID = IP Deskphone MAC address

Time To Live	Length = 2
	TTL= 180 [seconds]

End Of LLDPDU	Length = 0
Port Description	Length = 15 Port Description = “ IP Deskphone”
System Description	Length = the length of the system description string System Description = “IP Deskphone, xxx, Software: 0604D97” where: xxx = 1220, 1230 Software = software version. “0604D97” is an example only.
System Capabilities	Length = 4 System capabilities = 0x24 [Deskphone + Bridge] Enabled capabilities = 0x24 If you disable the PC Ethernet port, the advertised enabled capabilities configured to IP Deskphone only.
Port VLAN ID	PVID = 0 The IP Deskphone does not support port-based VLAN operation.
Port And Protocol VLAN ID	PPVID = 0 Port and Protocol VLAN is not supported and not enabled.
VLAN Name	VLAN name field is configured to “data” and “voice”.
Protocol Identity	1. STP: Protocol identity = the first 8 bytes of an STP PDU starting with the Ethertype field. Length = 8 Protocol Identity = 0x00 0x26 (type/length field of Ethernet packet, size=38) 0x42 0x42 0x03 (LLC header indicating STP)

TLV	Fields
	0x00 0x00 (Protocol Identity field from STP BPDU) 0x00 2. 802.1x: Length = 3 Protocol identity = 0x888E—(802.1x Ethertype) 0x01—(Version field from 802.1x frame) 3. LLDP: Length = 2 Protocol identity = 0x88CC—(LLDP Ethertype)
MAC/PHY Configuration/Status	Auto-negotiation support/status = Bit 0 = 1 [Auto-negotiation supported] Bit 1 = 1 or 0, depending on the current auto-negotiation status, for example, either enabled or disabled.
	PMD auto-negotiation advertised capability = 0x4000 - 10BASE-T half duplex mode 0x2000 - 10BASE-T full duplex mode 0x0800 - 100BASE-TX half duplex mode 0x0400 - 100BASE-TX full duplex mode
	Operational MAU Type = 10 – UTP MAU, 10BT, half duplex mode 11 – UTP MAU, 10BT, full duplex mode 15 - 2-pair Category 5 (CAT5) UTP, 100BT, half duplex mode 16 - 2-pair CAT5 UTP, 100BT, full duplex mode
Power Via MDI	MDI power support = 0: Bit 0 = 0 – Powered Device Bit 1 = 0 – PSE MDI power not supported Bit 2 = 0 – PSE MDI power state disabled Bit 3 = 0 – PSE pair selection can not be controlled
	PSE power pair = 1 Power Class = 2 for 1220/1230 IP Deskphones Power Class = 3 for 2007/1120E/1140E/1165E IP Deskphones
Link Aggregation	Aggregation status = 0; the link is not capable of being aggregated, and currently is not in aggregation. Aggregated Port ID = 0
Maximum frame size	The MAC/PHY supports an extension of the basic MAC frame format for Tagged MAC frames. The maximum frame size is configured to 1522.
LLDP-MED System Capabilities	Bit 0 = 1—LLDP-MED Capabilities–supported Bit 1 = 1—Network Policy–supported Bit 2 = 1—Location Identification–supported

TLV	Fields
	Bit 3 = 0—Extended Power using MDI-PSE–not supported Bit 4 = 1—Extended Power using MDI-PD–supported Bit 5 = 1—Inventory–supported The Class Type field can be configured to 3 -Telephone
Network Policy Discovery	Application Type-1—voice Unknown Policy Flag (U)—1 only if the policy is unknown Tagged Flag (T)—configure accordingly Reserved (X)-0 VLAN ID—configure accordingly L2 Priority—configure accordingly DSCP Value—configure accordingly
Location Identification Discovery	Coordinate-based LCI–16 bytes Civic Address LCI I–variable length This format can have more than one address element and one address element can range from a minimum of 7 to 256 bytes. ECS ELIN I–variable between 10 and 25 bytes Although location is received, it is not available to end user in this release of the SIP Software.
Extended Power-via MDI Discovery	Power Type = 01–PD Device Power Source = 00–Unknown. There is no hardware support for determining the power source. Power Priority = 0010–High Power Value = Maximum power required as shown below:
	1120E NTYS03 = 8 1140E NTYS05 = 8 1165E = 1220 = 1230 =
Software Revision	Configure to the software version being used
Manufacturer Name	“avaya-xy”, where: xy is a 2-digit manufacturer code as shown below:
	1220: Code 04 1230: Code 04
Model Name	Contains a string, which specifies the IP Deskphone model, for example, “IP Deskphone xxx”, where, xxx is one of the following values: 1220, 1230

DHCP

There are two methods of provisioning DHCP.

1. No DHCP (Manual configuration): All the Device Settings parameters are configured manually on the IP Deskphone with SIP Software.
2. Yes: The IP Deskphone with SIP Software is configured to get a standard set of Device Settings parameters from the DHCP server.

NO DHCP mode

No DHCP mode is also known as Manual Device Settings. In this mode, the IP Deskphone does not need a DHCP server because no DHCP requests are sent during startup. All necessary parameters must be configured manually or through the device configuration file.

The minimum following parameters must be configured to achieve normal operation:

- SET IP
- Net Mask
- Gateway
- DNS IP
- Provisioning server IP, protocol, and, if user authentication is required to access the provisioning server, the user credentials

  Note: TFTP protocol does not require user authentication. Device ID and password are ignored. FTP protocol requires user authentication and the default Device ID is anonymous with no password.

  HTTP protocol can operate with or without user authentication. If no authentication is required, make sure to clear the Device ID and password fields in the configuration dialogue.

SET IP

Select SET IP in the Device Settings menu to configure the IP address.

Net Mask

Enter a subnet mask In the Net Mask field.

Gateway

Enter an IP address of the local gateway in the Gateway field.

DNS IP1 and DNS IP2

Configure the Domain Name Servers (DNS) IP addresses DNS IP1 and DNS IP2.

Ntwk Port Speed

There are three options to provision the network port speed.

- Auto – Link speed is auto negotiated with the network device and attached PC.
- 10BT – Link speed is available for up to 10 Megabit Full Duplex on the network and the PC port.
- 100BT – Link speed is available for up to 100 Megabit Full Duplex on the network and the PC port.

Ntwk Port Duplex

There are three options available to provision the Network Port Duplex for Network Port Speed of 10BT or 100BT.

- Auto – duplex is autonegotiated. Avaya recommends that Auto Negotiate mode is used on the network and the IP Deskphone.
- Force Full – duplex is forced to FULL. Use Force Full mode only when the network is forced Full Duplex. Otherwise a duplex mismatch results.
- Force Half – duplex is forced to HALF. Use Force Half mode only when the network is forced Half Duplex. Otherwise a duplex mismatch results.

Disable Voice 802.1Q

If 802.1Q is disabled, standard Ethernet frames are transmitted. If 802.1Q is enabled, all frames transmitted by the Ethernet driver have the 802.1Q tag bytes inserted between the source MAC address and the protocol type field.

Voice VLAN

Configure the Voice VLAN.

Table 15: Telephony Port (incoming)

Voice VLAN Setting	Result
No VLAN (default setting)	All telephony traffic that is transmitted on the telephony port is forwarded untagged.
Manual (value from 1 to 4094)	All telephony traffic that is transmitted on the telephony port has an 802.1q header appended and its Voice VLAN ID is set to the value manually configured here. Enter a value from 1 to 4094.

VLAN Filter

Configure the VLAN Filter (not available if Voice VLAN is configured as No).

Table 16: VLAN Filter

VLAN Filter Setting	Result
Enabled (box is checked)	Traffic is forwarded to the IP Deskphone port, based on a review of the MAC address of the packet as well as the 802.1q tag value. Traffic is forwarded through the IP Deskphone port, (to the network stack of the IP Deskphone), only if the packet matches the MAC address of the IP Deskphone and contains the Voice VLAN tag. If the Automatic VLAN Discovery feature is used, the filter is adjusted dynamically as the IP Deskphone validates the VLAN suggested by DHCP.
Disabled	The VLAN filter is disabled by default. If the VLAN filter is disabled, traffic is forwarded to the IP Deskphone port based only on a review of the MAC address of the packet. The IP Deskphone accepts traffic addressed to the MAC address of the IP Deskphone as well as any broadcast or multicast packets from any VLAN.

Ctrl Priority bits

802.1Q priority bits for the control or signaling stream. There is not a control priority bit prompt if 802.1Q is not enabled.

Media Priority bits

802.1Q priority bits for the media (audio) stream There is not a media priority bit prompt if 802.1Q is not enabled.

Disable PC Port

With a disabled PC Port, the IP Deskphone with SIP Software does not receive or send packets to or from the PC port. No device can use the PC port to connect to the network.

Data VLAN

Configure the Data VLAN (the VLAN applicable to the PC port). The behavior of the PC port is summarized in the following table.

Table 17: PC Port (incoming traffic from PC)

Data VLAN	Result
No VLAN	All traffic received on the PC port is forwarded based on the MAC address. The packets are not modified in any way.
Manual (value from 1 to 4094)	All untagged packets received from the PC have an 802.1q header appended and the VLAN ID is configured to the value that is manually provisioned in this field. Any packet arriving on the PC port that is already tagged with matching VLAN ID is forwarded as is. Packets tagged with different VLAN IDs are dropped. Enter a value from 1 to 4094.

Table 18: PC Port (outgoing)

Data VLAN	Result
No	All traffic received on the network port and telephony port is forwarded to the PC port based on MAC address only. The packets are not modified in any way.
4095 (Note that the value 4095 is not a valid VLAN ID. This is intentional to ensure a proper VLAN ID is entered.)	Traffic is forwarded to the PC port based on a review of the MAC address of the packet, as well as the value configured in the Data VLAN field. Traffic is forwarded out the PC port (to the PC) only if the packets contain the Data VLAN tag. Untagged traffic and traffic with a VLAN tag other than the Data VLAN are dropped.

Data Priority bits

There is not a prompt for data priority bits if 802.1Q is not enabled.

PC-Port Untag all

Configure PC-Port Untag-All, which configures PC-Port VLAN Tag Stripping.

When enabled, all outgoing traffic to the PC has the VLAN tag removed. When disabled, packets are sent to the PC as is if Data VLAN is configured as No or if Data VLAN is enabled and the VLAN tag matched. Packets with different VLAN ID are discarded.

Cached IP

Leave unchecked to conform to the DHCP standard and to obtain an IP address from the DHCP server. Only check Cached IP to force the IP Deskphone to start with a cached IP address in the event that the IP Deskphone cannot connect to the DHCP server and obtain an IP address.

Port Speed and Duplex

In the Network menu, Auto Negotiation mode is the default setting for initial startup. Typically, the IP Deskphone is connected to a network that supports Auto Negotiation, and the IP Deskphone selects the best speed and duplex mode available. There is no intervention required under normal operation.

Important:

Avaya recommends that Auto Negotiation mode is used on the network and the IP Deskphone. Use Full Duplex mode only when the entire network is running in Full Duplex mode. Otherwise, a duplex mismatch results.

If the IP Deskphone is connected to a network configured for Full Duplex mode only, the IP Deskphone cannot automatically negotiate the proper configuration. Therefore, in this instance, to allow the IP Deskphone to work at the optimum speed and duplex mode, Full Duplex mode must be enabled.

Ignore GARP

Gratuitous Address Resolution Protocol (GARP) Protection prevents the IP Deskphone with SIP Software from GARP spoof attacks on the network.

The IP Deskphone with SIP Software provides the ability to ignore GARP messages.

Provisioning

The next four menu items apply to configuring the provisioning parameters.

Server URL

Enter the numeric IP address, name, or URL of the provisioning server in the Server URL box.

Protocol

Select the protocol used to access the provisioning server, either TFTP, FTP, or HTTP.

Device ID

Enter the Device ID (User ID) used by the provisioning server for authentication of the IP Deskphone with SIP Software.

Password

Enter the password used by the provisioning server for authentication of the IP Deskphone with SIP Software.

PVQMon IP

If the Proactive Voice Quality Monitor (PVQMon) server is available on the network, enter the PVQMon IP address.

NAT Traversal

The next five menu items apply to configuring the NAT Traversal Method if the IP Deskphone with SIP Software is connected to the network through a Network Address Translation device or a firewall.

NAT Signal

The IP Deskphone with SIP Software supports two methods of NAT traversal of signaling path: SIP_PING and STUN.

The default NAT traversal method is None.

SIP_PING is a legacy Avaya Proprietary protocol for NAT Traversal for SIP signaling only. STUN is an Internet standard for NAT traversal.

If the value for NAT traversal is not configured as None, this parameter overrides the value of the parameter SIP_PING specified by the device configuration file for NAT_SIGNALLING.

If the value for NAT traversal is set to None, the value of SIP_PING, if specified by the device configuration file for NAT_SIGNALLING, is used instead.

NAT_SIGNALLING is required for networks that use STUN or SIP_PING for NAT traversal.

NAT Media

The IP Deskphone with SIP Software supports STUN for media path the NAT traversal. The NAT Media feature can be disabled by setting the NAT_Media field in Device Settings menu to NONE.

Note:

STUN protocol cannot coexist with Application Layer Gateway (ALG), Media Portals, or RTP Proxy servers. If STUN is selected, ensure none of these devices are configured in the SIP Proxy server.

NAT TTL

Enter the Time to Live (TTL) in seconds for the NATport if STUN is enabled. The IP Deskphone with SIP Software pings the open ports on an interval less than the TTL to prevent the NAT from tearing down the ports.

STUN S1 IP

Enter an IP address for the STUN S1 IP device. The STUN server must reside in the public Internet for STUN protocol to be effective.

STUN S2 IP

Enter an IP address for the STUN S2 IP device. The STUN server must reside in the public Internet for STUN protocol to be effective.

Configure the device settings

This section describes how to provision the **Device Settings** parameters, regardless of DHCP mode (Yes or No DHCP). Parameters that are not needed in Yes DHCP mode are grayed out on the IP Deskphone screen and cannot be modified.

Perform the following procedure to provision the **Device Settings** parameters on the IP Deskphone with SIP Software.

Provisioning the Device Settings parameters

1. Press the **Globe/Services** key on the IP Deskphone quickly twice. The **Network** menu appears:

   Table 19: Network menu

1. Server Settings
2. Device Settings
3. Diagnostics
4. Lock

2. To navigate in the Network menu, use the **Navigation key cluster**:

   Table 20: Using the Navigation key cluster to navigate in the Network menu

Key	Action
Down	Moves highlight down
Up	Moves highlight up
Right	Selected current menu item
Left	Closes menu
Select key (center of cluster)	Selects current menu item

   Menu items can also be accessed through the IP Deskphone keypad. Press the number (1 to 4) corresponding to a menu item to highlight that menu item. Press the **Select** key to select the highlighted menu item.

   To close this menu, use the **QUIT** key.

3. In the Network, choose **2. Device Settings**.
4. Enter the admin password. Press the **OK** context-sensitive soft key.
5. Use the **Navigation key cluster** to edit an item:

   Table 21: Using the Navigation key cluster to navigate in the Device Settings menu

Key	Action
Down	Opens list or moves highlight down list.
Up	Moves highlight up list.

Key	Action
Right	Selects the next item or moves the cursor right in an edit item.
Left	Selects the previous item or deletes a character in the edit field.
Select key (center of cluster)	Selects the highlighted item in the combo box or ends edit mode.

In Edit mode, the first field of the item is highlighted. Press the **Apply&Reset** context-sensitive soft key to save settings in the Device Settings menu after all the necessary changes are made. This action resets the IP Deskphone.

Press the **Return** context-sensitive soft key to exit the Device Settings menu without saving any changes.

6. Navigate to the first item in the Device Settings menu to configure **EAP**.

   Table 22: EAP configuration

Enable 802.1x (EAP): □
Device ID:
Password:

   A check mark appears in the check box if the item is active.

7. If the **Enable 802.1x (EAP)** check box is checked, fill in the **Device ID** and the **Password**.

8. Configure **802.1ab Link Layer Discovery Protocol Data (LLDP)** in the **Enable 802.1ab (LLDP)** menu.

   Use the right arrow in the **Navigation key cluster** to highlight the **Enable 802.1ab (LLDP)** box.

   A check mark appears in the check box if the item is active.

   Table 23: 802.1ab configuration

Enable 802.1ab (LLDP): □

9. Configure **DHCP** in the Device Settings menu.

   Use the right arrow in the **Navigation key cluster** to highlight the **DHCP** box. Press the **Select** key to access a list of DHCP mode choices:

   - Yes
   - No

   Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select No.

Table 24: DHCP configuration

DHCP :		▼	
	No		
	Yes		

10. If DHCP is disabled, select **SET IP** in the Device Settings menu to configure the IP address.

    Table 25: SET IP configuration

SET IP : 0.0.0.0

11. If DHCP is disabled, configure **Net Mask and Gateway**. Enter a subnet mask in the **Net Mask** field and enter the IP address of the local gateway in the **Gateway** field.

    Table 26: Net Mask and Gateway configuration

Net Mask : 0.0.0.0
Gateway : 0.0.0.0

12. If DHCP is not Full DHCP, then to configure the DNS enter the IP address of the local DNS servers DNS IP1 and DNS IP2.

    Table 27: DNS configuration

DNS IP1 : 0.0.0.0 DNS IP2 : 0.0.0.0

13. Configure **Ntwk Port Speed** in the Device Settings menu.

    Use the right arrow in the **Navigation key cluster** to highlight the **Ntwk Port Speed** box. Press the **Select** key to access a list of Ntwk Port Speed choices:

    - Auto
    - 10BT
    - 100BT

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select a Ntwk Port Speed parameter.

    Table 28: Ntwk Port Speed configuration

Ntwk Port Speed :		▼	
	Auto		
	10BT		
	100BT		

14. Configure **Ntwk Port Duplex** in the Device Settings menu.

    Use the right arrow in the **Navigation key cluster** to highlight the **Ntwk Port Duplex** box. Press the **Select** key to access a list of Ntwk Port Duplex choices:

    - Auto
    - Force Full
    - Force Half

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

    Table 29: Ntwk Port Duplex configuration

Ntwk Port Duplex :		▼	
	Auto		
	Force Full		
	Force Half		

15. **Disable Voice 802.1Q** .

    Press the **Select** key to disable Voice 802.1Q.

    A check mark in the **Disable Voice 802.1Q** box indicates that Voice 802.1Q is disabled.

    Table 30: Voice 802.1Q configuration

Disable Voice 802.1Q: □

16. Configure **Voice VLAN**.

    Use the right arrow in the **Navigation key cluster** to highlight the **Voice VLAN** box. Press the **Select** key to access a list of Voice VLAN mode choices.

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Voice VLAN:	No VLAN	▼	

17. Enable or disable the **VLAN Filter**.

    Press the **Select** key to enable the VLAN Filter.

    A check mark in the **VLAN Filter** box indicates that the VLAN Filter is enabled.

    Table 31: VLAN Filter configuration

VLAN Filter : □

18. If 802.1Q is enabled, configure **Ctrl Priority bits**.

Use the right arrow in the **Navigation key cluster** to highlight the **Ctrl Priority bits** box. Press the **Select** key to access a list of Ctrl Priority bits choices.

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 32: Ctrl Priority bits configuration

Ctrl Priority bits :	Auto	▼	
	Auto		
	Value from 0 to 7		

19. If 802.1Q is enabled, configure **Media Priority bits**.

    Use the right arrow in the **Navigation key cluster** to highlight the Media Priority bits box. Press the **Select** key to access a list of Media Priority bits choices.

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 33: Media Priority bits configuration

Media Priority bits :	Auto	▼	
	Auto		
	Value from 0 to 7		

20. **Disable PC Port** configuration.

    Press the **Select** key to turn off the PC port on the IP Deskphone.

    A check mark in the **Disable PC Port** box indicates the PC port is disabled.

    Disable PC Port can be used in an environment where administrators do not want users accessing the data network through the built-in PC port.

Table 34: Disable PC Port configuration

Disable PC Port : □

21. If the PC port is not disabled, configure **PC Port Speed** in the Device Settings menu.

    Use the right arrow in the **Navigation key cluster** to highlight the PC Port Speed box. Press the **Select** key to access a list of PC Port Speed choices:

    - Auto
    - 10BT
    - 100BT

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select a PC Port Speed.

Table 35: PC Port Speed configuration

PC Port Speed :	Auto	▼	
	Auto		
	10BT		
	100BT		

22. If the PC port is not disabled, configure **PC Port Duplex** in the Device Settings menu.

    Use the right arrow in the **Navigation key cluster** to highlight the PC Port Duplex box. Press the **Select** key to access a list of PC Port Duplex choices:

    - Auto
    - Force Full
    - Force Half

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select PC Port Duplex.

Table 36: PC Port Duplex configuration

PC Port Duplex :	Auto	▼	
	Auto		
	Force Full		
	Force Half		

23. **Disable Data 802.1Q**.

    Press the **Select** key to disable Data 802.1Q.

    A check mark in the **Disable Data 802.1Q** box indicates that Data 802.1Q is disabled.

Table 37: Data 802.1Q configuration

Disable Data 802.1Q: □

24. Configure the **Data VLAN**.

    It is possible to enable or disable separate VLANs for anything other than voice and signaling.

    Use the right arrow in the **Navigation key cluster** to highlight the Data VLAN box. Press the **Select** key to access a list of Data VLAN mode choices:

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 38: Data VLAN Configuration in Device Settings menu

Data VLAN :	LLDP VLAN NAME	▼	
	No VLAN		
	LLDP VLAN NAME		
	Value from 1 to 4094		

If 802.1ab (LLDP) is disabled, Data VLAN cannot be automatically configured. LLDP VLAN NAME does not appear in the in the list of Data VLAN mode choices.

25. If 802.1Q is enabled, configure **Data Priority bits**.

    Use the right arrow in the **Navigation key cluster** to highlight the **Data Priority bits** box. Press the **Select** key to access a list of Data Priority bits choices:

    Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 39: Data Priority bits configuration

Data Priority bits :	Auto	▼	
	Auto		
	Value from 0 to 7		

26. Configure **PC-Port Untag-All**, which configures PC-Port VLAN Tag Stripping. Check the check box by pressing the **Select** key in the Navigation key cluster to enable this item.

    If the check box is not checked, tag stripping is disabled and the packet is sent to the PC port unmodified. If the check box is checked, tag stripping is enabled and the 802.1q header is removed (assuming one exists) from the packet before it is forwarded through the PC port.

    If Data VLAN is enabled in step 15, PC-Port Untag All is enabled and the check box is checked by default. In this case, the outgoing tag is stripped. Override the default by pressing the **Select** key to remove the checkmark.

    If Data VLAN is disabled, PC-Port Untag All is disabled and the check box is not checked by default. In this case, the ingress tag is not stripped. The default can be overridden by pressing the **Select** key, which places a checkmark in the box.

Table 40: PC-Port Untag-All Configuration in Device Settings menu

PC-Port Untag all : □

27. Configure **Cached IP**.

    Leave the check box unchecked to conform to the DHCP standard and to obtain an IP address from the DHCP server. Only check the Cached IP check box to force the IP Deskphone to start with a cached IP address if the IP Deskphone cannot connect to the DHCP server and obtain an IP address.

To force the IP Deskphone to start with a cached IP address, press the **Select** key.

A check mark in the **Cached IP** box indicates that the IP Deskphone is forced to start with a cached IP address.

Table 41: Cached IP configuration

Cached IP: □

28. Configure the **Ethernet port mode**.

    Ignore GARP:

    GARP requests handling. The default can be overridden by pressing the **Select** key, which places a check mark in the box.

    Table 42: Ethernet port mode configuration

Ignore GARP: □

29. Enter the Provisioning parameters: Server URL, Protocol, Device ID, and Password.

    The next four tables show menu items in the Device Settings menu that apply to configuring the Provisioning parameters.

    - **Server URL**:

      Enter the URL of the provisioning server in the **Server URL** box.

      Table 43: Server URL configuration

Provisioning		
	Server URL:	0.0.0.0

    - **Protocol**:

      Select either TFTP, FTP, or HTTP used for provisioning in the **Protocol** box.

      Table 44: Protocol configuration

Protocol:	TFTP	
	TFTP	
	FTP	
	HTTP	

    - **Device ID**:

      Enter the **Device ID** used by the provisioning server for authentication of the IP Deskphone.

Note:

TFTP does not require a Device ID. If FTP or HTTP are used, enter the information for a Device ID

Note:

Enter your User ID as the Device ID.

Table 45: Device ID configuration

Device ID :	XXXXXXXXX	

- **Password**:

  Enter the **Password** used by the provisioning server for authentication of the IP Deskphone.

  Note:

  TFTP does not require a Password. FTP and HTTP do require a password.

Table 46: Password configuration

Password :	XXXXXXXXX	

30. Configure the **PVQMon IP** address.

    If this server is available on the network, enter the **PVQMon IP** address.

Table 47: PVQMon IP configuration

PVQMon IP :	0. 0. 0. 0	

31. Configure the NAT Traversal Method. Use one of the following methods to traverse Network Address Translation (NAT) devices for the IP Deskphone with SIP Software.

    The next five tables show menu items in the Device Settings menu that apply to configuring the NAT Traversal Method.

    - **NAT Signal**

      Select either None, SIP_PING, or STUN to define the NAT traversal mode for SIP signaling.

Table 48: NAT Signal configuration

NAT Traversal	NAT Signal :	

		None
		SIP_Ping
		STUN

- **NAT Media**

  Select either None or STUN to define NAT traversal mode for media.

  Table 49: NAT media configuration

NAT Media :	STUN	
	None	

- **NAT TTL**:

  Enter the Time to Live (TTL) in seconds for the NAT traversal.

  Table 50: NAT TTL configuration

NAT TTL :	000	

- **STUN S1 IP**:

  Enter an IP address for the STUN S1 IP device.

  Table 51: STUN S1 IP configuration

STUN S1 IP :	00. 00. 00. 00	

- **STUN S2 IP**:

  Enter an IP address for the STUN S2 IP device.

  Table 52: STUN S2 IP configuration

STUN S2 IP :	00. 00. 00. 00	

Chapter 12: Multiple Appearance Directory Number (Single Call Arrangement)

The Multiple Appearance Directory Number (MADN) (Single Call Arrangement) feature operates differently depending on the type of Communication Server. For instance, the MADN feature operates differently on the Communication Server 2000 than the Communication Server 1000. For more information about the MADN feature and how it operates on the Communication Servers, see the following sections:

- Communication Server 2000 and Communication Server 2100 on page 127
- Communication Server 1000 on page 128

Communication Server 2000 and Communication Server 2100

The MADN feature allows a Directory Number (DN) to appear on more than one IP Deskphone with SIP Software. The MADN with Single Call Arrangement (SCA) feature allows multiple IP Deskphones to appear as a single line to a caller. Any one of the IP Deskphones in a group with MADN can initiate or answer a call, but only one call can be active at any given time. Any other user in the group can join the active call by picking up the handset of the IP Deskphone.

With the MADN SCA feature configured on multiple phones of different registered SIP users, the phones share one single DN. An incoming call to this DN causes all the phones in the group to ring.

Any user of an IP Deskphone with the MADN SCA feature can put a call on hold or can prevent others from joining in the active call.

If a user's group is active (as seen by line icon being off-hook) and the user picks up the handset, the user is automatically joined to an ongoing MADN call (unless the server restricts this feature for privacy or other factors).

Vertical services

Vertical services are CS 2000 and CS 2100 features that can be activated or deactivated by dialing a defined code; for example, Privacy. Even though no more than one active session can be established for the MADN SCA group, members of the group can still enter certain vertical services.

Currently, the available vertical service is Privacy.

Privacy

A user can activate the privacy service by putting the current session on hold and dialing the privacy code. The CS 2000 connects the IP Deskphone to the Media Application Server (MAS) to hear a confirmation for its request and terminates the session. The user takes the original session off hold.

Privacy access codes

The privacy access codes are: PRV, PRLA, PRLC. For example: PRV = 191 PRLA = 192 PRLC = 193 If the initial state of the MADN group is nonprivate , the PRV access code is used to toggle between privacy on and privacy off. If the initial state of the MADN group is private, the PRLA access code allows bridging and PRLC closes it.

Feature dependencies and restrictions

The minimum release to support MADN SCA feature is 1.1. Multi-User login is not supported.

Communication Server 1000

The Multiple Appearance Directory Number (MADN) allows a Directory Number (DN) to appear on more than one IP Deskphone with SIP Software. The MADN with Single Call Arrangement (SCA) feature allows multiple IP Deskphones to appear as a single line to a caller. Any one of the IP Deskphone phones in a group with MADN can initiate or answer a call, but only one call can be active at any given time.

With the MADN SCA feature configured on multiple phones of different registered SIP users, the IP Deskphones share one single DN. An incoming call to this DN causes all the IP Deskphones in the group to ring.

Any other user in the group can join the active call by accessing the line key with SCA provisioned.

Any user with the MADN SCA feature can put a call on hold. Any other user in the group can pickup the held call by accessing the line key with SCA provisioned.

The state of the user’s group is reflected in the line key icon. Three states are available, idle, active and held.

No specific provisioning is required on the IP Deskphone with SIP Software. This feature will be automatically operational if it is enabled on the server.

Chapter 13: Multiuser

The Multiuser feature allows multiple SIP user accounts to be in use on the IP Deskphone at the same time. Multiple users, each with their own account, can share a single IP Deskphone allowing each user to receive calls without logging off other users. One user can have multiple user accounts (for example, a work account and a personal account) active at the same time on the same IP Deskphone. You can register each account to a different server, and for each account, the IP Deskphone exposes the functionality available to that account.

One account is considered a primary account and is used by default for most IP Deskphone operations. Each account is associated to a line key; the primary account is always on the bottom right line key of the IP Deskphone, and an arbitrary key (including a key on an Expansion Module) can be selected for additional accounts.

You can use the line key to do the following:

- start dialing
- place a call using the corresponding user account
- to answer an incoming call targeted to that account

Initiating a call without pressing a line key (for example, by dialing digits at the idle screen and lifting the handset) uses the primary account.

A running IP Deskphone is associated to a single profile that represents one configuration of the IP Deskphone with all relevant persistent data such as preferences and call logs. A different profile is associated to each account used as a primary account. The IP Deskphone can store up to five different profiles; the IP Deskphone takes data from the profile associated to the current primary account. A number of configurations are independent of profiles and tied directly to an account making them available to that account regardless of the primary account you use (for example, voice mail ID).

The IP Deskphone receives and answers calls targeted at any of the registered accounts; the incoming call screen indicates who the call is for. You can place an outgoing call using any of the accounts; the account that you use is displayed on the dialing screen. When a call is active, information from both local and remote parties appear on the screen.

Regardless of which account receives the call, incoming call logs, outgoing call logs, and instant messages appear in a single list. The IP Deskphone indicates the local user in the detailed view of the entry.

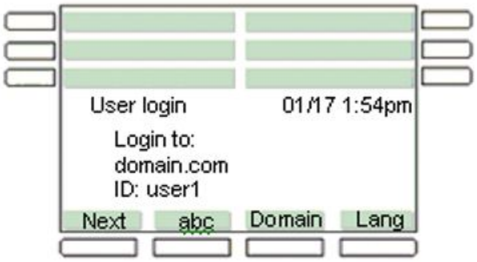
Some features are only available to the primary account, such as instant messaging, retrieving parked calls by token, and establishing ad-hoc conference calls.

If you log off of the primary account, the IP Deskphone unregisters all other accounts at the same time. These accounts are registered automatically after you log on the primary account (it is possible to use a different primary account to log on) again. When the IP Deskphone restarts, all accounts that were logged in before the IP Deskphone restarted, are automatically logged back on. The provisioning server can also configure the users who are allowed to log on to the IP Deskphone.

Navigation

Initial logon

To logon for the first time, you must enter a user name and password, and specify if the logon is permanent or not. On the logon screen, you can choose which domain you want to access, and change the language you want to use. You can use the Domain key only to select a domain from the configured list; you cannot modify domains.


User login 01/17 1:54pm

Login to:
domain.com
ID: user1

Figure 13: Primary logon screen

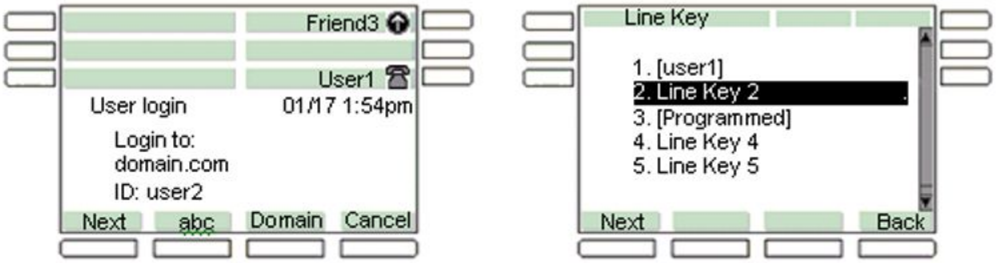
After you log on, the idle screen appears on the IP Deskphone. If there is no profile for the primary account, the IP Deskphone automatically creates a profile. You can create up to five profiles. If you exceed the limit of five profiles, the IP Deskphone automatically deletes the least recently-used profile.

Similarly, configurations for each user of the primary account are loaded after a user logs on to the IP Deskphone. The configurations are independent of the profile; if the account you use is registered as the secondary account (not the primary account), the IP Deskphone uses the configurations of the primary account. The IP Deskphone keeps up to 24 sets of configurations (one set for each user). If you exceed the limit of 24 sets of configurations, the IP Deskphone automatically deletes the least recently-used set, and a new account is registered.

Additional logons

The Login command in the System menu allows you to register additional accounts. If you log on as a secondary user, you cannot change the language selection.

The following figure shows the secondary logon screens.


User login 01/17 1:54pm

Login to:
domain.com
ID: user2

Line Key

1. [user1]
2. Line Key 2
3. [Programmed]
4. Line Key 4
5. Line Key 5

Figure 14: Example of secondary logon screens

You can specify a Line Key for a new account. By default, the IP Deskphone selects the first unused key. If the IP Deskphone reaches the configured limit on concurrent logons and you select the Login command, an error message appears.

During the logon operation, a `login in progress` message appears on the IP Deskphone screen. The IP Deskphone can receive calls for user accounts that are registered; however, other features are not available until the logon process is complete. The IP Deskphone does not display a profile selection prompt and does not create a profile for the secondary account.

Automatic logon

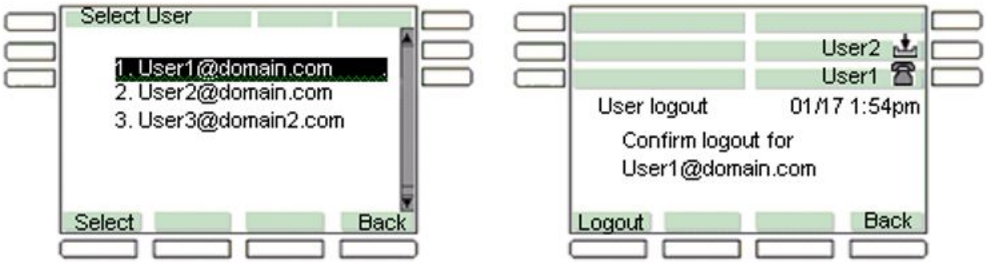
If you are logged on when the IP Deskphone is switched off, the IP Deskphone automatically logs you back on when you restart the IP Deskphone. If multiple users are logged on when the IP Deskphone is switched off, the IP Deskphone automatically logs all users back on, one after the other, when you restart the IP Deskphone.

If the automatic logon feature is disabled in the device configuration file, automatic logon does not occur.

You can provision the IP Deskphone with user credentials to automatically logon multiple accounts on the initial startup. See Primary account logout on page 135.

Logging off

The Logout command in the System submenu, prompts you to select an account, asks for confirmation, and then proceeds to log off the account. Logging off an account frees the corresponding Line key and does not require a password.


Select User

1. User1@domain.com
2. User2@domain.com
3. User3@domain2.com

User logout 01/17 1:54pm

Confirm logout for User1@domain.com

Figure 15: Example of log off screens

Primary account logout

Logging off the primary account causes all other accounts to log off automatically and the IP Deskphone to display the logon screen. The IP Deskphone logs back in the secondary accounts automatically after you register a new primary account or the same primary account.

If you restart the IP Deskphone after you logged off the primary account, the logon screen appears on the IP Deskphone. Logging on a new primary account leads to automatic logon of the secondary accounts.

The list of programmed feature keys is part of the IP Deskphone profile. Logging off one primary account and logging on a different account can change the set of feature keys. If a secondary account is assigned to a key that is also in the new set of feature keys, the secondary account takes precedence; the secondary account is logged on and the feature key acts as a Line key. If the account is logged off manually, the programmed feature key becomes available.

Secondary account logout

If you log off a secondary account by selecting the secondary account in the Logout Select User screen, the IP Deskphone removes the secondary account from the autologon list. After you restart the IP Deskphone, the IP Deskphone does not logon the secondary account.

Server failover

If the connection to your account proxy is lost, the IP Deskphone notifies your account and periodically attempts to reconnect. Some features, such as incoming calls, remain accessible for other accounts, but other features are not available until connection is reestablished or you cancel the reconnection. Cancelling the connection to your account is the same as logging off. If you are using the primary account, the IP Deskphone returns you to the initial logon screen. If you are using a secondary account, that secondary account is removed from the list of secondary accounts that are logged on automatically.

If more than one account loses connection, the IP Deskphone attempts to reconnect to each account in sequence. The IP Deskphone tries to reconnect the first account to lose connection until that account reregisters or you cancel the attempt. Then the IP Deskphone attempts to reconnect the next account that lost connection. Cancelling the reconnection of the primary account immediately abandons reconnection of all other accounts, logs off secondary accounts that are still connected, and returns the IP Deskphone to the logon screen.

The IP Deskphone uses a single logon queue for automatic logons and failover. This means that if automatic logons are still pending when an account cannot connect, a reconnection attempt for that account can only begin after all automatic logons are complete or cancelled.

Cable unplugged

If the IP Deskphone detects that the network cable is unplugged while accounts are logged on, the IP Deskphone assumes that all accounts have lost their connection to the server. When the cable is reconnected, the IP Deskphone proceeds to reregister all accounts, starting with the primary account.

Line keys

Each registered user is associated to a separate line key. Each line key displays the name of the registered account and some basic state information for the account.

The primary account is associated to the first bottom-right line key of the IP Deskphone. If you are using a secondary account, the order of the next available line key is from bottom to top and right to left on the IP Deskphone, followed by the keys on the Expansion Module from bottom to top and right to left. You can select a different available line key for secondary accounts during the logon process.

The following figure is an example of the IP Deskphone with and Expansion Module and multiple accounts.

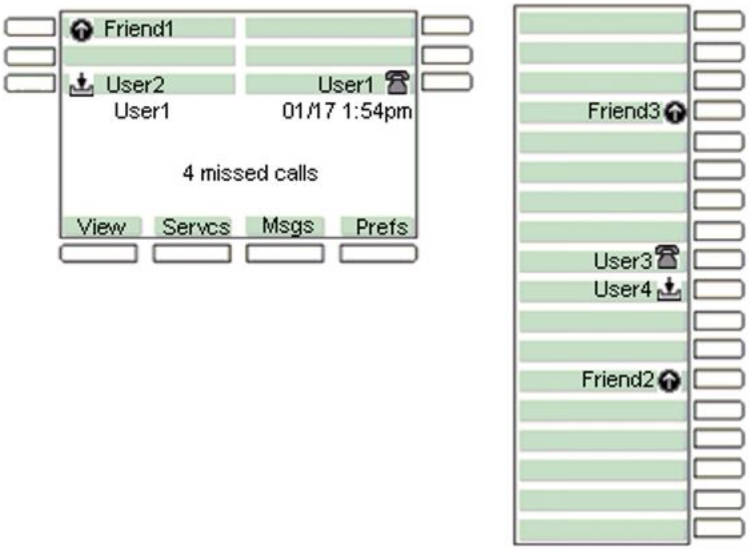


Figure 16: IP Deskphone with Expansion Module and multiple accounts

Pressing a line key brings up a dialing prompt, initiates a call to a preselected target, or answers an incoming call. See Making a call on page 137.

At select account prompts, such as the Logout screen or User Settings screen, pressing a line key highlights the corresponding account. See Account selection on page 147.

The icon for each line key reflects the state of the account associated with that line key.

- If there is a call for the account, a IP Deskphone icon displays the state of the call, such as when the call is on hold or is ringing.
- If there is more than one call, the state of the most active call is displayed.
- Missed incoming calls and new voice mail messages for the account are indicated with an icon. The icon supplements the `NN missed calls` message on the idle screen and the red LED which cannot provide per-account information.
- The MADN, do-not-disturb, and call forwarding features also affect the appropriate line key icon of the account.

Making a call

You can place a call using any of the registered user accounts. The account that you select determines:

- the proxy used
- the domain name used for the call target (if none was specified)

- the caller the target sees is calling
- the service-package-dependent features that are available

Receiving a call

When you receive an incoming call, the account that the call is intended for is displayed on the IP Deskphone. The line key of that account displays the icon for an incoming call. You cannot use a different account to answer the call.

If you are receiving multiple calls at the same time, a list of all active and incoming calls appears. If you select a specific call in the list, you can choose to answer or process that specific call. The IP Deskphone sorts the list by the most recent incoming call first. If there are numerous calls to process, you can configure the selected call to automatically select the last incoming call to make it easier to answer, or to leave the selected call static. The selected call does not jump as new calls come in, but remains on the same call, as new calls are added, to make it easier for the user to process that call.

If the calls are for different accounts, the line keys associates with the accounts receiving the incoming calls display an incoming-call icon.

Being in a call

When a single call is active, the screen displays the local account in use and the remote user. If multiple calls are active, each call appears on a single line. The local account for the active call appears on the context line. Each line key reflects the most active call state of the account the line key is associated with.

The active call is affected by operations such as transfer or call parking. One exception is the New Call action which uses the primary account by default, but can be overridden by pressing another line key to initiate a call.

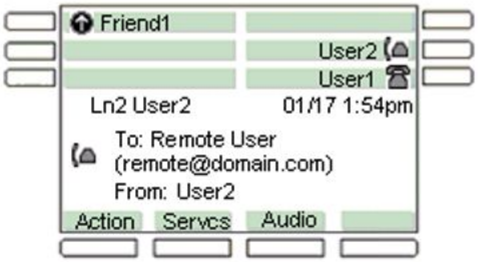
You can use your account to transfer or park an active call that is received on that account. The exception is the New Call action because it uses the primary account by default. You can override the New Call action by pressing another line key to initiate a call.

Joining calls into an ad-hoc conference always uses the conference server of the primary account. Calls that are on accounts that cannot access the server cannot be joined. After you create an ad-hoc conference, you can join additional calls into the same conference. You cannot create more than one ad-hoc conference at a time.

You can join any two calls with the 3-way call feature, regardless of the account. The service package of the account to which a call is associated determines which operations, such as

Call Park, are available on that call. After you establish a 3-way call, the join functionality becomes unavailable until the 3-way call is terminated.

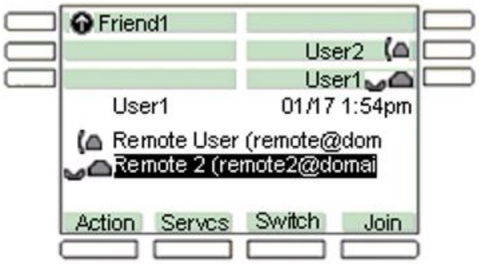
The following figure is an example of the IP Deskphone with one call.



Friend1
User2
User1
Ln2 User2 01/17 1:54pm
To: Remote User
(remote@domain.com)
From: User2

Figure 17: Example of the IP Deskphone with one call

The following figure is an example of the IP Deskphone with multiple calls.



Friend1
User2
User1
User1 01/17 1:54pm
Remote User (remote@dom
Remote 2 (remote2@domai

Figure 18: Example of the IP Deskphone with multiple calls

Instant messages

You can only receive or send instant messages from the primary account. Incoming messages for secondary accounts are rejected, are not displayed on the screen, and are not added to the instant message logs.

Menu features

The menus displayed on the IP Deskphone are customized to match the service package of the active account that is accessing the menu. Menus are accessed from the Idle screen when the primary account is active. For example, you can only use the **Retrve** context-sensitive soft key to retrieve a parked call if call parking is allowed by the service package of the primary user.

Similarly, accessing the Address Book through the **Directory** hard key displays the Address Book of the primary account. However, accessing the address book in select mode (for example, while dialing or selecting an item for a speed dial key) accesses the address book of the user account that is in use on the address-input screen.

Modifying settings

Preferences, such as Voice Mail and IM settings, are available for each account. The main **Preferences** menu includes a **User Settings** entry. If you select **User Settings**, you are prompted to select a registered account. After you select a registered account, a menu appears that lets you modify the settings of the account you selected.

Per-account call notification options

The **Call Settings** entry in the **User Settings** menu provides you with a number of configuration options relating to how incoming calls for a particular account are treated.

The configuration options are:

- what kind of audio alert you want to use (ring tone, beep, or nothing)
- whether you want the red LED to blink
- whether you want the call to be added to the Incoming Call logs

IM Settings

IM Settings is located in the **User Settings** menu. Any change in settings on the primary account takes effect immediately. You can also modify settings for a secondary account, but the modifications do not take effect until you register the secondary account as the primary account.

Voice Mail settings

Voice Mail Settings is located in the **User Settings** menu. You can program different voice mail addresses and IDs for each account. To access the voice mail of a secondary account, press the line key of the secondary account to obtain a dial prompt, and then press the **VMail** context-sensitive soft key.

Waiting messages are reported in the following two ways:

- The red LED lights up if any account has a waiting message.
- A shaded envelope icon appears on the line key of each account that has a waiting message (unless the account is in a call).

Remembering settings after logout

The IP Deskphone remembers up to 24 sets of configurations for each profile. If you configure settings for an account and you log off the account, the settings are restored after you log back onto the account (as either a primary account or a secondary account).

If you log on an account that you did not save the settings in a profile for, the IP Deskphone creates a new set of default settings for that account. If there are already 24 sets of configurations in the profile, the IP Deskphone discards one set that is not currently registered with the account, and replaces the discarded set with the new set that is saved in the account profile.

Programmable keys

You cannot use a line key associated with a registered account for programmable features. The Program Key screen lists all the line keys associated to an account. If you select a line key associated to an account, an error message appears.

The Do Not Disturb, Call Forward, and Presence keys are associated to a specific user account that you create, and determine which account status to affect. See User status on page 143..

By default, pressing a Speed Dial programmed key initiates a call using the primary account. If you press a line key to obtain a dialing prompt, and then press a speed dial key, the IP Deskphone uses the account associated with that line key. When accounts are registered on different domains, you can program and use speed dial keys with targets that are only reachable on the domain of a secondary account.

Important:

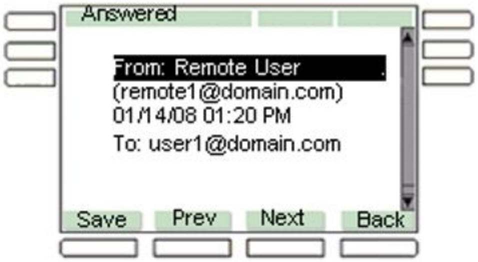
The Speed Dial keys always use the primary account to determine the presence state of the target.

The Instant Message keys always use the primary account, because IM support is disabled for secondary accounts.

Inbox, outbox, and instant message log

Each profile has a single inbox, a single outbox and a single instant message log. The detailed view of the call log entry indicates the local account associated to each entry; that is, the source of outgoing calls and the target of received call.

The following figure is an example of the Inbox call details view.



Answered

From: Remote User
(remote1@domain.com)
01/14/08 01:20 PM
To: user1@domain.com

Figure 19: Example of the Inbox call details

Call logs and IM logs provide many ways of initiating a call to the address identified by the selected entry, such as lifting the handset. In most cases, the primary account is used. However, if you press a line key to initiate the call, the call uses the account associated with the line key.

If call logs and IM logs are invoked in the **selection** mode, you cannot initiate a call directly because the **Select** context-sensitive soft key populates a dial prompt or other input field with the selected target. The operation already in progress determines which account you can use.

For example:

If you press the line key to obtain a dial prompt, then press the **Inbox** key to select a target, press **Select**, and then press **Send**; the line key that you originally pressed determines the account you can use.

Address Books

Each registered account can have a network-based address book. Each profile contains a local address book that is independent from all network address books.

Accessing the Address Books, by pressing the Directory hard key from the Idle screen, displays the address book of the primary account. If the primary account does not have a network address book, the local address book is accessed.

Accessing the Address Book in Selection mode always accesses the address book of the current account. For example, after obtaining a dial prompt by pressing Line Key 2, you can press the Directory key to access the address book of the account associated to Line Key 2. You can access the network-based directory of the appropriate account if it is available; otherwise, the IP Deskphone accesses the local address book.

You can only access the network-based address book of secondary users in selection mode. You cannot modify the address book of a secondary account on the IP Deskphone. However, modifications that you make to the address book remotely, such as using a different client of the Personal Agent, are reflected on the IP Deskphone.

The local address book is shared by all accounts that do not have a network-based address book. You can modify the local address book if the primary account does not have a network-based address book. Changes to the network-based address book of the primary account are not reflected in the local address book.

If you use the Friends view, you can always access and modify the address book of the primary account (local or network-based). There is no selection mode for the Friends view. You can only monitor and view the presence information of Friends of the primary account in the Friends view.

User status

The features associates with the User status include the following:

- Do Not Disturb
- Call Forwarding
- Presence

Do Not Disturb

Selecting the **Do Not Disturb (DND)** command from the Services menu prompts you to specify which account you want to place in the DND mode. The option **all** allows you to place all accounts in the DND mode (the all option is highlighted by default). By selecting an option, the IP Deskphone prompts you to confirm the operation before proceeding.

Activating DND for a specific account automatically causes calls to that account to be rejected with a busy signal. However, the IP Deskphone can still receive calls to other accounts. After DND mode is active for an account, the label of the account line key periodically displays a DND indicator.

The following scenarios apply to DND.

- If you select a single account that is in DND mode, the IP Deskphone displays a prompt that asks if you want to deactivate the DND mode.
- If you select a single account that has Call Forwarding active, an error message appears to indicate that DND cannot be activated.
- If you select the option **all**, and at least one account is not in DND mode, DND mode is activated for all accounts. If an account is in Call Forward mode, Call Forward is disabled.
- If you select all and all accounts are in DND mode, DND mode is deactivated for all accounts.

If you use a programmed DND feature key, the account that is affected by the DND feature key is determined when the feature key is configured. After you press the DND feature key, the IP Deskphone behaves as described in the preceding scenarios, except that there is no confirmation prompt displayed. The IP Deskphone performs the operation immediately, and a message appears to indicate what was done.

The DND mode for each account is persistent. If you restart the IP Deskphone, or log off the account and log the account back on, the account maintains the original state.

Call Forwarding

After you select the **Call Forward** command from the **Services** menu, the IP Deskphone prompts you to specify the account that you want to place in Call Forward mode. The option **forward all** places all accounts in Call Forward mode in one operation, and the option **forward none** deactivates Call Forward for all accounts at the same time.

The following scenarios apply to Call Forward:

- If you activate call forwarding for a specific account, the IP Deskphone automatically redirects all calls to the selected account to the address that you specify. The target address must be reachable from the domain of the account. Other accounts can still receive calls. The line key label periodically indicates that Call Forward mode is active.
- If you select a single account that does not have Call Forward or DND active on it, the IP Deskphone prompts you to specify a forwarding target, and the mode you select is then enabled. If DND is already active, a message appears indicating that Call Forward cannot be activated. If Call Forward is already active, a message appears asking you if you want to deactivate Call Forward.

- If you select the **forward all** option, all accounts are in Call Forward mode using the provided target, and DND is deactivated for all accounts. If accounts are already in Call Forward mode for a different target, the accounts are updated to use the new target.
- If you select the **forward none** option, the Call Forward feature is deactivated for all accounts for which the Call Forward feature is currently active.

After you press a single account Call Forward programmed key:

- If the account is already forwarding calls to the programmed target, call forwarding is deactivated.
- If the account is not forwarding calls to the programmed target, the account is set to forward calls to the given target, disabling DND if necessary, and overriding any other call forward target that is active for the account.

After you press a forward all programmed key:

- If all accounts are already set to forward calls to the key target, call forward is disabled for all accounts (behaves like the **forward none** option).
- If all accounts are not configured to forward calls to the key target, call forwarding is activated for all accounts using the key target (behaves like the **forward all** option).

If you do not perform any Call forwarding or DND operations, you can press the **single** and **all** keys to switch one or all accounts between **forwarding to key's target** and **not forwarding** states.

The Call Forward mode and target is persistent for each account. If you restart the IP Deskphone, or log off the account and log the account back on, the account maintains the original state.

Presence

After you select the **Presence** command from the **Services** menu, you are prompted to specify which presence state of the account you want to modify. The option **all** lets you set all accounts to the same presence in one operation.

If you select a single account, the current state of the account is displayed. You can change the current state of the account by entering the new presence state and note. After you confirm the operation, the new presence state is applied.

If the **all** option is selected, no current state is displayed, and you are immediately prompted to select the new state. The new state is applied to all registered accounts.

If you use a programmed Presence feature key, the account that is impacted by the Presence feature key is determined after the feature key is configured.

After you press a **single account** Presence programmed key:

- If the account is already set to the programmed presence state, the account is set back to the **Connected** presence state.
- If the account is not already set to the programmed presence state, the account is set to the programmed presence state.

After you press the **all accounts** Presence programmed key:

- If all accounts are already set to the programmed presence state, all accounts are set to the **Connected** presence state.
- If all accounts are not already set to the programmed presence state, all accounts are set to the programmed presence state.

As like the Call Forwarding keys, if you do not perform any Presence operation, you can use the **single** and **all** keys as toggles. However, the presence states are not entirely under your control. Some states are applied automatically (for example, On The Phone), and all states are applied by sending a message to the SIP proxy which can choose to not accept the change. As a result, it is possible for a **set all presence** operation to not configure all accounts to the programmed presence; if you press the Presence key again, another attempt is made to apply the programmed presence to all accounts. It is more effective to program a separate Presence key to set all accounts to the Connected state.

Events that update presence states automatically occur for each account. For example, the **On The Phone** state is applied to any account that has at least one call active.

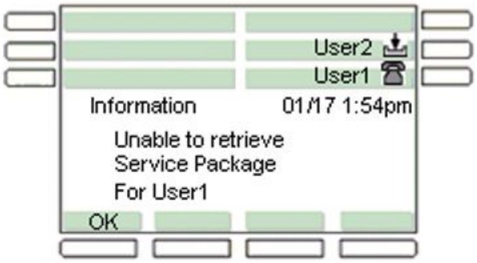
Account presence is not retained after logging off or restarting the IP Deskphone.

Notifications

The IP Deskphone can spontaneously display messages on the screen to report events that you did not initiate. This includes events such as failure to retrieve a service package and availability of a new location list.

These spontaneous notifications do not indicate which account is affected by the event. A message appears to indicate the affected account.

The following figure is an example of an account notification.



Unable to retrieve
Service Package
For User1

Figure 20: Example of an account notification

It is possible for the same event to occur for multiple accounts at the same time, or in quick succession. In the preceding figure, the accounts are displayed one after the other.

Account selection

There are a number of scenarios where you are prompted to select an account (for example, logoff, per-account settings, programming keys).

The scenarios fall into the following two categories:

- Prompts where you must select exactly one account. If only one account is logged on, the prompt does not appear. The IP Deskphone selects the single account automatically, and immediately displays the next screen. Otherwise, the primary account is always at the top of the list, and is highlighted when the prompt first appears.
- Prompts where an all or none option is available.

Pressing an account line key highlights the corresponding item in the account list. If no selection is made in a certain amount of time, the prompt acts as if you pressed the Back Context-sensitive soft key, canceling whichever operation required selection of an account.

Feature dependencies and restrictions

The number of line keys on the IP Deskphone limits the number of accounts that you can register simultaneously. The IP Deskphone is limited to six accounts. Connecting an Expansion Module to the IP Deskphone increases the limit by 18, allowing for 24 registered accounts. Additional Expansion Modules do not increase the limit further.

These are hard limits. Further restrictions may be imposed by the administrative policy. See Configuration files on page 148.

Configuration files

You can configure the parameter, MAX_LOGINS in the device configuration file. The number that you configure determines the maximum number of user accounts that you can log on at the same time. Numbers higher than the number of line keys on the IP Deskphone are equivalent to no limit other than the line key. The default is unlimited. A value of 1 allows a single user at a time. A value of 0 is treated the same as a value of 1 because it is not possible to restrict the IP Deskphone to 0 logons.

The number of concurrent logons can never exceed 24, regardless of the value of MAX_LOGINS.

The parameter SELECT_LAST_INCOMING determines which call is selected when there are multiple calls ringing (or active). The default value is 0. If the value is 0, the first selected call is left static as new calls come in or are dropped. . If the value is changed to 1 then the selected call in the call list jumps to the most recent ringing call when it is added to the list.

Performance characteristics

Because the multiuser feature can allow the IP Deskphone to have multiple users logged on to the IP Deskphone at the same time, the chances of numerous multiple calls increase. The IP Deskphone can handle five simultaneous incoming calls at a time without any noticeable impact. But as the number of simultaneous incoming calls increase, there is a noticeable delay in ringing and updating the display to present all the calls to the user. It may take up to five seconds for 10 simultaneous incoming calls, and this time increases as the IP Deskphone receives more simultaneous incoming calls.

Chapter 14: Features

This chapter contains the following topics:

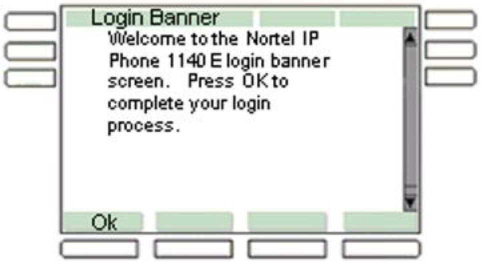
Customizable banner for login

SIP Software allows the IP Deskphone to display a customizable banner when you log on to the IP Deskphone. When the login banner is provided with login banner text and is configured as "enable", the IP Deskphone displays the banner text on the screen when the user logs on.

The banner text is only displayed in the language that is provisioned (changing the IP Deskphone configured language does not change the banner text language). The banner appears only for the primary user of the IP Deskphone. In a multiuser configuration, a secondary user logon does not cause the banner to appear, even if the login banner is configured as enable.

If the login banner is configured a enabled, the banner screen on the IP Deskphone is displayed after the final step of the logon process.

The following image is an example of the Login Banner screen which displays the provisioned banner text.



Login Banner

Welcome to the Nortel IP Phone 1140 E login banner screen. Press OK to complete your login process.

Ok

Figure 21: Login Banner

The following table describes the function of the context-sensitive soft key for the Login Banner screen.

Table 53: Context-sensitive soft key for the Login Banner screen

Context-sensitive soft key	Action
Ok	Completes the login process and dismisses the login screen.

The following table describes the function of the Navigation keys for the Login Banner screen.

Table 54: Navigation

Key	Action
Up and down arrows	Allows you to scroll up and down the banner text.
Left and right arrows	No action (the text is word-wrapped automatically).
Enter	No action.

The following table describes the outside actions on content for the Login Banner screen.

Table 55: Outside actions on content

Key or action	Result
Inbox	No action.
Outbox	No action.
Directory (Address book)	No action.
Goodbye	No action.
Expand (IM Box)	No action.
Copy	No action.
Services	Press once, no action. Press twice invokes the Network menu.
Quit	No action.
Headset	Brings up the dial prompt (in case the user wants to place an emergency call).
Hold	No action.
Dialpad	No action.
Handsfree	Brings up the dial prompt (in case the user wants to place an emergency call).

Key or action	Result
Off Hook	Brings up the dial prompt (in case the user wants to place an emergency call).
Mute	No action.
Volume up and volume down	No action.
User-defined feature keys	No action.
Incoming call	Incoming calls get a Do Not Disturb (DND) response while the banner is displayed.

The user must explicitly dismiss the banner screen (like a location list), and the IP Deskphone goes in DND mode until the banner is dismissed. The IP Deskphone cannot make or receive any calls, other than an emergency call, until the banner is dismissed.

If any other pop up messages or prompts, such as a location prompt, occur while the banner is displayed, the pop up messages or prompts appear below the banner screen, and are viewed by the user only after the user dismisses the login banner.

The following configuration flag is used for enabling or disabling the customized login banner.

LOGIN_BANNER_ENABLE Y/N (Default: N)

The banner text is defined in a separate text file that is linked from the original configuration file.

The banner text file is a separate file downloaded by provisioning. The banner text file is specified much like the current dialing plan is specified (file name listed in 12xxSIP.cfg, under section [LOGIN_BANNER]), and is downloaded when enabled or disabled.

To be accepted, the file must contain at least one byte and must be no larger than 2048 bytes. The encoding of the file must be UTF-8, or compatible with UTF-8, to ensure that all the characters are displayed properly.

Speed Dial List

When configured by provisioning, a feature key can be used as a "Speed Dial List". The feature key and the contents of the Speed Dial List must be specified by the provisioning mechanism. The user cannot modify or delete the feature key used by the Speed Dial List and cannot modify the content of the Speed Dial List.

Invocation of the Speed Dial List is similar to any other feature key invocation. The Speed Dial List key causes a full screen list to appear on the IP Deskphone and the user can automatically dial one of the offered choices.

The contents of the Speed Dial List can vary (context-sensitive) based on the current call state of the IP Deskphone and the type of Speed Dial List entry configured. Only entries in the Speed

Dial List can be context-sensitive; not all speed dial keys or individual features keys are context-sensitive.

A Speed Dial key, or one included in a Speed Dial List, can cause any call that it placed on hold (when invoked) to be unheld automatically when the call completes, based on a new value that must be configured when a Speed Dial key is created or configured.

Administration and use of the Speed Dial List feature

Provisioning the device configuration provides the IP Deskphone with the following features:

- Index of key to use as Speed Dial List. You can use the following flag to disable the Speed Dial List feature by configuring the key index to less than two (2).

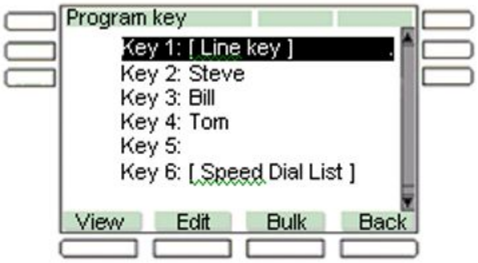
  SPEEDLIST_KEY_INDEX <key index>
- Label to use for the Speed Dial List key.

  SPEEDLIST_LABEL <text>

The IP Deskphone retrieves the device configuration through provisioning, and if the SPEEDLIST_KEY_INDEX flag is configured to a valid programmable key that can be used for the feature (greater than one (1) and less than or equal to the available number of programmable keys), the following events occur:

1. The IP Deskphone checks for a previously loaded "Speed Dial List" file (a file containing the contents of the speed dial list), which must be properly configured and uploaded to the IP Deskphone through provisioning.
2. The IP Deskphone parses the file, and configures the feature key specified by SPEEDLIST_KEY_INDEX to hold the Speed Dial List.
3. If the key defined for use by the Speed Dial List is already in use, the defined key is overwritten and is assigned Speed Dial List functionality.
4. The Speed Dial List feature key uses the label that is provided in SPEEDLIST_LABEL, and cannot be modified by the end user.

The following screen describes the feature key used by the Speed Dial List in the feature key programming interface.



Program key

Key 1: [Line key]
Key 2: Steve
Key 3: Bill
Key 4: Tom
Key 5:
Key 6: [Speed Dial List]

Figure 22: Main feature key programming screen showing Speed Dial List provisioned on key 6

A feature key provisioned for use as a Speed Dial List has a similar appearance to all other programmed feature keys on the idle screen (or in-call screen). The label used for that key is provided through provisioning.

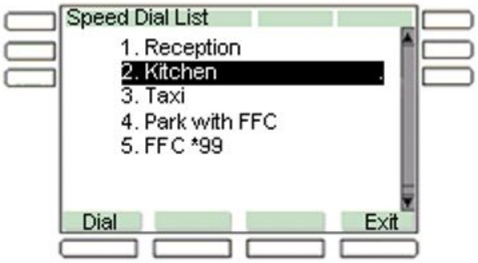
When the user presses the feature key provisioned as a Speed Dial List, the list of speed dials configured appears on the screen, and the user can select an item from the list to invoke Speed Dial.

If the Speed Dial List is empty, or ends up empty due to context-sensitive hiding of contents, the error message, "Not available", is displayed on the screen with a "Dial List" context line.

Speed Dial List screen

The Speed Dial List screen for the IP Deskphone is where the user can select or invoke one of the provisioned Speed Dial List entries.

The following image is an example of the screen that appears after the user presses the feature key that is provisioned as the Speed Dial List for the IP Deskphone.



Speed Dial List

1. Reception
2. Kitchen
3. Taxi
4. Park with FFC
5. FFC *99

Figure 23: Example of a Speed Dial List

The Speed Dial List screen displays all the Speed Dial List entries provisioned for the user. The listed items displayed are based on the provisioned list as well as the current Idle or Mid-call state of the IP Deskphone. When the Speed Dial List is invoked while the IP Deskphone is idle, only Speed Dial List entries that are configured as IDLE are displayed. Similarly, only items marked as MID CALL are displayed if the Speed Dial List is invoked while the IP Deskphone is in a call.

The following table describes the function of the context-sensitive soft keys for the Speed Dial List screen.

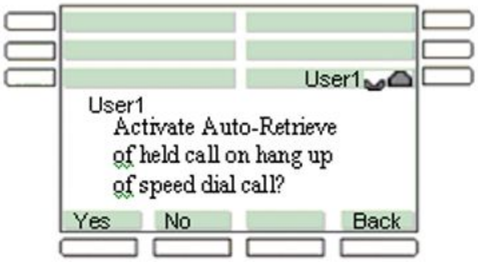
Table 56: Context-sensitive soft keys for the Speed Dial List screen

Context-sensitive soft key	Action
Dial	Invokes the selected speed dial.
Exit	The screen is dismissed without invoking a Speed Dial List entry.

Auto retrieve flag

Because the auto retrieve behavior is added to the regular speed dial keys (programmed keys) instead of just speed dial list entries, the auto retrieve flag is configured for programmed speed dial keys.

The following screen appears as the last step, after the "Enter Subject" prompt, in the creation or modification of a Speed Dial key to allow the user to configure the auto retrieve behavior for the Speed Dial function.



User1

User1

Activate Auto-Retrieve
of held call on hang up
of speed dial call?

Figure 24: Speed Dial Key creation — last step

The following table describes the function of the context-sensitive soft keys for the Auto Retrieve screen.

Table 57: Context-sensitive soft keys for the Auto Retrieve screen

Context-sensitive soft key	Action
Yes	Enables the Speed Dial Auto Retrieve behavior.
No	Disables the Speed Dial Auto Retrieve behavior.

Context-sensitive soft key	Action
Back	Dismisses the screen and returns you to the previous key programming screen.

If the auto retrieve behavior is enabled on a Speed Dial key (programmed keys) or Speed Dial List entry that is invoked, and a call is placed on hold to invoke the current key or entry, the IP Deskphone attempts to remove the call on hold after the key or entry call is complete.

The following is a description of how the auto retrieve function operates.

1. A is talking to B when A invokes the Speed Dial List and selects an entry.
2. The call between A and B is placed on hold, and A places another outgoing call to C (a URI specified in the Speed Dial List entry).
3. When the call between A and C is complete, if the auto retrieve flag is enabled for the Speed Dial, then the IP Deskphone attempts to take the call between A and B off hold.

   If another call comes in during the call between A and C, or if the state of the call between A and B changes when the call between A and C is active, the re-connection of the call between A and B may not always happen.

The following is an example of a Speed Dial List file that must be loaded through provisioning.

```
speedDialList.txt [key] label=S1 target=s1@avaya.com retrieve=YES
mode=MidCallOnly type=spdial [key] label=S2 retrieve=NO mode=IdleOnly
subject=subject2 target=s2@avaya.com type=spdial [key] label=S3
retrieve=NO target=s3@avaya.com type=spdial
```

Busy Lamp Field

The Busy Lamp Field (BLF) is an alternate presence monitoring mechanism for the IP Deskphone that allows presence functionality on proxies supporting BLF. The IP Deskphone subscribes to receive presence information for watched users through notifications. The BLF mechanism allows the user to subscribe to the proxy and receive the presence state list for all the users it is configured to watch. The provisioning of the proxy configures the watch lists for users.

The proxy involved supports BLF and has a mechanism for setting the lists of watchers and presentees. The UI is not available to a user to enable or disable this feature. If the IP Deskphone is provisioned for this feature, it makes use of it.

If enabled, and the proxy provides notifications to the user, the IP Deskphone uses the notifications to update any speed dial keys that receive presence information in the BLF notifications.

Configuration flags for Busy Lamp Field

The provisioning must provide a configuration flag, containing relevant URI, to the IP Deskphone in order to use the Busy Lamp Field (BLF).

The following table describes the configuration flags used to configure the BLF.

Table 58: BLF configuration commands

Configuration command	Description
BLF_ENABLE Y/N/SCS/SIPX (default: N)	Enables or disables the BLF. When BLF_ENABLE has the SCS or SIPX value, the BLF_RESOURCE_LIST_URI parameter is ignored and the IP Deskphone autogenerates an URI of the following format: `~~rl~C~<username>@<domain>`
BLF_RESOURCE_LIST_URI <BLF URI>	Configures the BLF resource list URI for the BLF feature. `<BLF URI>` is the server-provided URI used to subscribe to BLF notifications (for example, `blf-resource-list@as.avaya.com`).

Chapter 15: Hotline service

The Hotline service allows you to provision a SIP IP Deskphone to a Hotline Phone. From a Hotline Phone, you can automatically make a call to a designated number.

A Hotline Phone is a dedicated IP Deskphone that has only one target. You cannot make a call to any other destinations; even emergency calls, such as E911 are not permitted. A Hotline Phone does not know the Hotline target and relies on the server to replace the To field of all INVITE messages sent from the Hotline Phone with the Hotline target to complete the call.

Important:

You cannot place calls if the server is unavailable during an upgrade.

Making a Hotline call

A call to a Hotline target is automatically placed when an off-hook condition occurs, or when you press digits during idle on-hook, and then lift the handset.

Hotline Service allows only one hotline user to login to the Hotline Phone. The Multi-user Login feature is restricted to one user only.

Hotline service restrictions

Because the Hotline Phone is a dedicated IP Deskphone used only for Hotline service, certain features are restricted on the Hotline Phone.

The following is a list of features, on the IP Deskphone, that are restricted on the Hotline Phone.

- Call Transfer
- Call Forward
- Voice Mail
- Call Park
- Instant Messaging

- MLPP
- E911 call

The display of each feature that is restricted on the Hotline Phone is blocked.

Provisioning

Hotline Service configuration is obtained from the Hotline Service Enable parameter from the service package or the device configuration file. The service package takes precedence over the device configuration file.

Service Package

You can turn Hotline Service, on or off, through the Service Package or the device configuration file. If the Hotline Service Enable parameter from the service package is configured as true, the Hotline Service is enabled (available) from the service package.

Device configuration file

The IP Deskphone uses the configuration parameters for the Hotline Service to indicate if Hotline Service is available and if a hotline call is in progress.

The following table describes the two configuration parameters in the device configuration file for Hotline Service.

Table 59: Hotline Service configuration parameters

Parameter name	Description	Default
HOTLINE_ENABLE	Indicates if Hotline Service is enabled or disabled.	No (indicates that Hotline Service is disabled)
HOTLINE_URL	Used as To field of INVITE message by the SIP IP Deskphone to notify the Proxy Server that this is a call from a Hotline Phone. The HOTLINE_URL is not a real URL of the Hotline target. The IP Deskphone has no idea about the Hotline target.	Hotline

Parameter name	Description	Default
	The Proxy server replaces the To field of INVITE request message with a real Hotline target when it receives an INVITE request from the Hotline Phone.	

Chapter 16: Session Timer Service

The Session Timer for the Session Initiation Protocol (SIP) feature (RFC4028) allows the Avaya 1200 Series IP Deskphone to support a keep-alive mechanism for SIP sessions. SIP sessions are periodically refreshed by UPDATE requests (or re-INVITES for the IP Deskphones that do not support UPDATE). The UPDATE requests are sent during an active call to allow endpoints or proxies to determine the status of a SIP session.

The Session Timer Service contains the following elements:

- Session-Expires header
- Min-SE header
- response message (422—Session interval too small)
- tag (timer) for existing headers

The SIP IP Deskphone generates, processes and handles the SIP messages that include the preceding elements.

Session-Expires header

The SIP Session-Expires header delivers the Session-Expires interval and provides information about the entity performing the refreshes. A value of "uac" indicates that the originating endpoint performs the refresh; a value of "uas" indicates that the terminating endpoint performs the refresh. The session interval is the maximum amount of time that occurs between session refresh requests in a dialog box before the session times-out. The minimum for this field is 90 seconds; the recommended value is 1800 seconds (30 minutes).

Min-SE header

The Min-SE header indicates the minimum value for the session expiration in units of delta-seconds. When your make a call, the presence of the Min-SE header informs the terminating endpoint, and proxies, of the minimum value that the originating endpoints accept for the session timer duration in units of delta seconds. When present in a 422 response, the Min-SE header indicates the minimum session value the terminating endpoint accepts. When present in a request or response, the value of the Min-SE header is 90 seconds or more. If the Min-SE header is not present, the default value is 90 seconds. It is a configurable parameter.

Provisioning

The IP Deskphone uses the configuration parameters for the Session Timer Service to indicate if the Session Timer Service is available, and to configure the duration of the session timer.

The following table describes the five configuration parameters in the device configuration file for Session Timer Service.

Table 60: Session Timer Service configuration parameters

Parameter name	Description	Default value
SESSION_TIMER_ENABLE	Indicates if the session timer service is enabled or disabled. If configured as Yes, the Session Timer Service for the IP Deskphone is enabled, and the behavior of the IP Deskphone complies with RFC4028. If configured as No, the Session Timer Service is disabled.	Yes
SESSION_TIMER_DEFAULT_SE	Indicates the default session expiration in seconds. The Session-Expires header, in a request, informs the terminating endpoint and proxies of the Session-Expires interval value that the originating endpoint requires for the session timer duration, in unites of delta seconds.	1800
SESSION_TIMER_MIN_SE	Indicates the minimum session expiration in seconds.	1800
SET_REQ_REFRESHER	Indicates what refresher value is configured in the initial session request. Value 0 indicates that the refresher is omitted; value 1 indicates that the refresher is configured to UAC; value 2 indicates that the refresher is configured to UAS.	0
SET_RESP_REFRESHER	Indicates what refresher value is configured in the 200 OK response. Value 0 indicates that the refresher is omitted (only valid when SET_REQ_REFRESHER is not equal to 0); value 1 indicates that	2

Parameter name	Description	Default value
	the refresher is configured to UAS; value 2 indicates that the refresher is configured to UAC.	

Chapter 17: Emergency Services

Overview

You can use the Avaya 1200 Series IP Deskphone to make an emergency call to the Public Safety Answering Point (PSAP), from any screen, without a user logon. When you connect to the PSAP, the IP Deskphone conveys the caller's location information to the PSAP. If you are not logged on to the IP Deskphone and you pick up the handset or press the handsfree or headset button, the message "Emergency calls only" appears on the screen of the IP Deskphone.

If you hang up before the connection is established, the IP Deskphone goes back to the initial state. After the connection is established, the call can only be ended by the Public Safety Answering Point (PSAP). If you hang up, the IP Deskphone switches to loudspeaker. If the IP Deskphone is already on the loudspeaker mode, and you press the hang up button, nothing happens. The call is still connected and can only be disconnected by the emergency operator.

Emergency calls originate on the IP Deskphone and are completed by the Call Server. The Call Server communicates with the emergency network or emergency systems for routing, call control, and location information. Although the IP Deskphone allows the user to enter location information, this location information is not used by all Call Servers. Some Call Servers derive the location information based on the number and location databases. Characteristics of emergency calls and limitations of emergency calls using the IP Deskphone are as follows:

- Making calls without logging on is only allowed for emergency calls (according to the defined dialing plan).
- Transmission of the location information depends on M5T SIP Stack version 4.1 (because it must be able to transmit multiple MIME types).

Location information

Effective Emergency services also rely on accurate location information. The IP Deskphone supports the inclusion of the x-nt-location header in SIP messages that provide location information to a Avaya Call Server. The location information is selected by the user during registration, and must be correctly provisioned at the Call Server level (see appropriate NTP specific to your Call Server).

Dialing plan configuration

To allow operator control of disconnect during an emergency call, the IP Deskphone must identify an emergency call as soon as an emergency call is initiated. The IP Deskphone uses an emergency flag in the dialing plan to identify an emergency call. When the dialing plan detects that an emergency number is dialed, it automatically switches to operator controlled disconnect mode when the call is answered. The dialing plan can have multiple emergency numbers.

The following outline describes the format for the dialing plan rules.

1. The first part contains one or more patterns. The patterns are used to match against the dialed number. Multiple patterns are separated by the | character.
2. The second part contains the resulting string used in the dial step.
3. The third part defines the parameters used by the UA to trigger specific dialing actions. The following parameters are defined in the third part and are separated by the | character if both are used.
    - t=xxxx: timer to stop collecting digits or perform automatic dialing out after the user enters the first digit. The xxx is a decimal number for the timer value in msec. The default timer is used if the timer is not specified in the digit map.
    - emergency: if specified, special call features are enabled to handle the call as an emergency call.

The following is an example of an emergency flag in the dialing plan:

```
911|911#  && sip:user@911.com && t=1000|emergency
```

This feature requires configuring the values for additional variables in the IP Deskphone config file.

The following table describes the configuration values for the emergency dialing plan.

Table 61: E911 Configuration in the IP Deskphone Config file

E911_USERNAME	The emergency user name used for making an emergency call that does not require a logon. You must configure the proxy with the same emergency user name, otherwise, the emergency call fails.
E911_PROXY	Default emergency proxy. This variable must contain the value that matches the value defined by one of the following variables specified in the same config file:

	• SIP_DOMAIN1 • SIP_DOMAIN2 • SIP_DOMAIN3 • SIP_DOMAIN4 • SIP_DOMAIN5 If E911_PROXY does not match the value defined by these five variables, or the variable E911_PROXY is not defined, the value of SIP_DOMAIN1 is used as the emergency proxy.
E911_PASSWORD	The password for emergency username that is used for making an emergency call that does not require login. The proxy must be configured with the same password, otherwise the emergency call fails.
E911_TXLOC	The variable that describes location information that must be sent with the REGISTER SIP message, or with the INVITE SIP message.

Important:

You must add a set of numbers (regular expressions) marked as "emergency" to the IP Deskphone dialing plan. Only these numbers are allowed for emergency calls that do not require logon.

Feature impact on configuration tasks

1. Configuring the SIP Proxy
   - The IP Deskphone must have an emergency user in order to make an emergency call without a user logon.
   - The IP Deskphone must have the necessary configurations values for automatic REGISTER of the emergency user (if you choose this implementation method).
   - You must add the emergency user to the proxy.
2. Adding the emergency user to the IP Deskphone config file
   - The IP Deskphone must have E911_USERNAME, E911_PROXY, and E911_PASSWORD configured for making emergency calls.

- The IP Deskphone must have a specified proxy that contains a user record with the specified user name and password.
- The IP Deskphone must have these values for automatic REGISTER of the emergency user (if you choose this way of implementation).
- You must add specified variables to the IP Deskphone config file.

3. Adding an emergency number

    - You must specify an emergency number for emergency calls to:
        - define the numbers that you can use for an emergency call that does not require logging on.
        - trigger emergency functionalities, such as the inability of an emergency call originator to hold or hang up the call after the call is established.
    - You can only dial these numbers if there is no user log on (or the IP Deskphone is blocked).
    - You must add the emergency number to the dialing plan. The emergency flag is mandatory. For more information on the format for dialing plan rules, see Dialing plan configuration on page 166.

4. Configuring Layer 2 switch of the DHCP server to provide the IP Deskphone with location information

    - You must configure the Layer 2 switch or DHCP server to provide the IP Deskphone with location information.
    - You must provide the IP Deskphone with location information because the IP Deskphone sends it to the PSAP when making an emergency call.
    - You must configure the Layer 2 switch or DHCP server.

Network Element

Domain list configuration and proxy set up

- You must properly configure the domain list, and the active proxy must be correct, valid, and support current features.
- You must properly configure the proxy to support current features.
- The proxy must be able to transmit mixed MIME-types (for successful transferring of the location information).

Setup and configure Layer 2 switch or DHCP server to provide the IP Deskphone with location information

- You must properly configure the Layer 2 switch or DHCP server to provide the SIP IP Deskphone with location information through LLDP-MED or DNCP protocols respectively.
- You must properly configure the Layer 2 switch or DHCP server to provide the IP Deskphone with location information.

Configure the proxy with emergency user name and password

- You must have configuration access to the proxy to arrange for an emergency user (if this manner of implementation is chosen).
- The emergency user and password at the proxy side must be identical to the emergency user and password that every IP Deskphone is configured with. Otherwise, you cannot make an emergency call without logging on.

Characteristics of emergency calls

The user does not have to log on to make an emergency call.

During an emergency call, the user:

- cannot make outgoing calls.
- is not notified of incoming calls and cannot accept incoming calls. Incoming calls receive a call waiting tone.
- cannot transfer, join, or conference the emergency call, place the emergency call on hold, or park the emergency call.
- cannot receive an incoming call.
- cannot auto-retrieve a parked call and auto-retrieval of parked calls is not displayed.
- cannot disconnect the emergency call. Only an emergency center or operator can disconnect the emergency call. If the user attempts to disconnect after the call has been made, the IP Deskphone switches to loudspeaker. If the loudspeaker mode is already on, the connection remains.
- cannot change Audio Quality.
- can reply to IM pop-ups, which are operational during an emergency call.

During an emergency call, the keys function as follows:

- the Services, Inbox, Outbox, and Address-Book keys are all disabled.
- a right click of the mouse does not show the services menu.
- the conspicuous keys are disabled.

- the mute key and hold key are disabled.
- the increase and decrease volume keys remain functional.
- the feature keys are visible and all except the speed-dial keys are functional.

Shut down and restart

If the IP Deskphone turns on or off, the IP Deskphone restarts in the usual way, reads the config file, and receives the location information through LLDP-MED or DHCP protocols (from the Layer 2 switch or DHCP server, which must be available and properly configured). Otherwise, the IP Deskphone is not provided with valid location information and cannot transmit that information when making an emergency call.

Chapter 18: IP Deskphone restrictions

Service package restrictions

Individual features and feature restrictions are sent to the IP Deskphone as a part of the service package every time a particular user logs on to the IP Deskphone. If the Call Server does not support service packages, or if the Call Server restricts some of the features in the service package, functionality of some features is restricted.

If functionality is restricted, the associated buttons and Context-sensitive soft keys are not accessible or do not respond.

Distinctive Ringing feature

The IP Deskphone does not support the CS 2000 and CS 2100 Distinctive Ringing feature.

Chapter 19: NAT firewall traversal

The objective of putting devices behind a Network Address Translator (NAT) is to protect the devices from external interruption and to extend the public IP address space. However, the shield to stop unsolicited incoming traffic also has the drawback of breaking a number of IP applications, including SIP.

If a device is behind a NAT, transport addresses obtained are not publicly routable, and therefore, not useful in a number of multimedia applications. The limited lifetime of the NAT port mapping can also cause the SIP signaling to fail. If a port mapping is idle, it can be released by the NAT and reassigned to other applications.

The STUN protocol lets an IP Deskphone discover the presence and type of NATs between the IP Deskphone and the public Internet. In addition, an IP Deskphone can discover the mapping between the private IP address and port number and the public IP address and port number. Typically, a service provider operates a STUN server in the public Internet, with STUN-enabled IP Deskphones embedded in end-devices, which are possibly behind a NAT.

A STUN server can be located using DNS SRV records using the domain of the service provider as the lookup. STUN typically uses the well-known port number 3478. STUN is a binary encoded protocol with a 20-octet header field and possibly additional attributes. The STUN protocol learns the public IP addresses, and therefore, some security is necessary.

To initiate a STUN lookup, the IP Deskphone sends one or more Binding Request packets using UDP to the STUN server. These packets must be sent from the same IP address that the IP Deskphone uses for the other protocol, because this is the address translation information that the IP Deskphone tries to discover.

The server returns Binding Response packets, which tell the IP Deskphone the public IP address and port number from which it received the Binding Request. The IP Deskphone knows the private IP address and port number it used to send the Binding Request, and therefore, it learns the mapping between the private and public address space being performed by the NAT. If the Binding Response packets indicate the same address and port number as the request, the IP Deskphone knows no NATs are present.

The IP Deskphone supports two methods for NAT traversal of the signaling path:

- SIP_PING
- STUN

The NAT traversal method can be selected manually through the Device Settings menu or configured through the device configuration file. The default NAT traversal method is NONE.

The IP Deskphone can conduct SIP dialogs through a Symmetric NAT using UDP. This allows the IP Deskphone to work from behind and/or in front of a symmetrical NAT with servers and/or clients that support RFC3581. For this feature to work properly, the receiving end device must support RFC3581. This feature is enabled or disabled through the USE_RPORT parameter in the device configuration file.

Note:

RFC3581 does not address NAT traversal for media or voice.

Chapter 20: Three-port switch and VLAN functionality

System overview

The Full VLAN support feature can create Avaya 1200 Series IP Deskphone Voice-VLAN and PC Data-VLAN on the three-port switch manually and automatically (see Figure 25: Voice-VLAN and Data VLAN on page 176).

If both Data and Voice VLANs are enabled on a three-port switch, only the frames with Data and Voice VLAN tagged go to networks. The IP Deskphone receives only the frames with Voice VLAN tagged and sends the frames with Voice VLAN tagged, while PC or Local Networks receive all kinds of frames.

When only voice VLAN is enabled on three-port switch, all kinds of frames go to the Network, the IP Deskphone receives only the frames with Voice VLAN tagged and send all frames with Voice VLAN tagged. PC or Local Networks receive all kinds of frames.

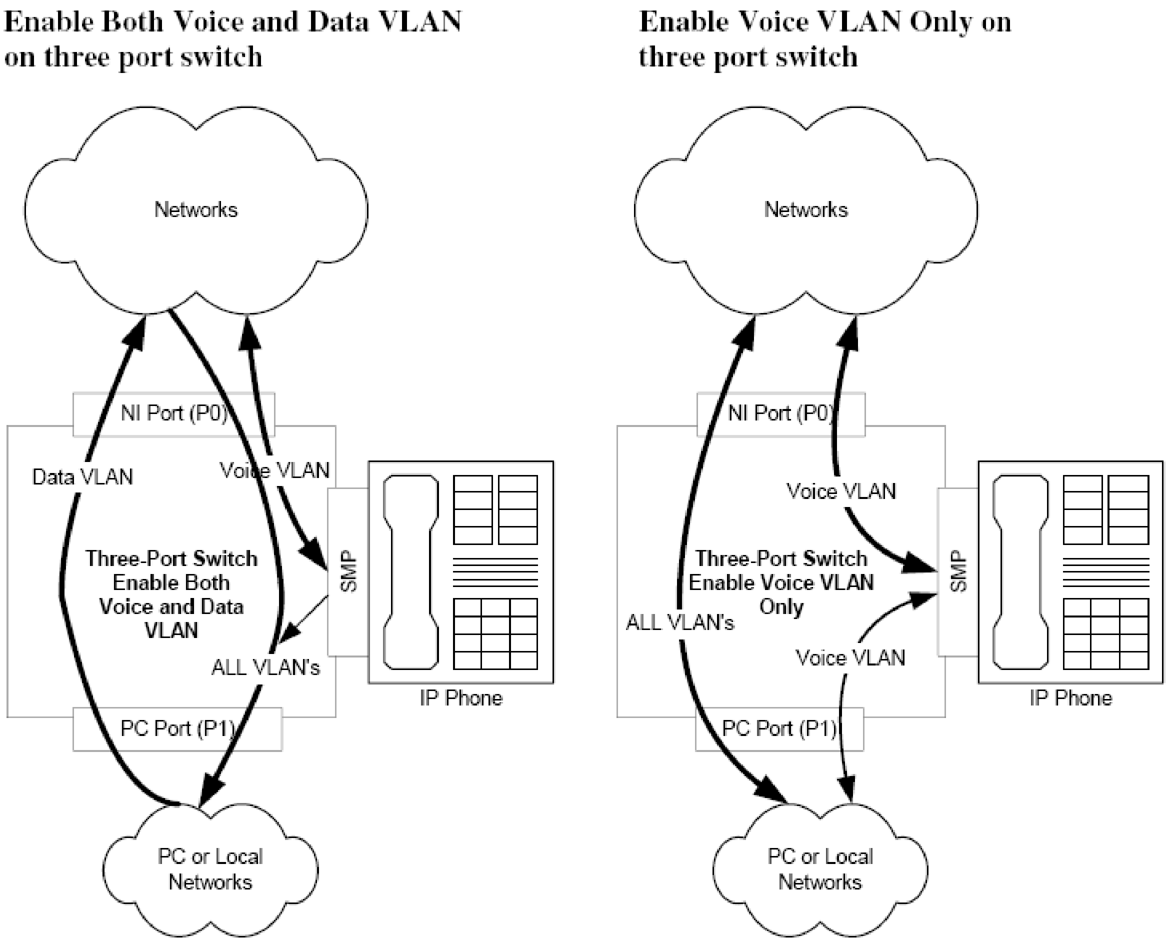


Figure 25: Voice-VLAN and Data VLAN

Table 62: Port functions on the three-port switch when VLAN is enabled

Ports	Voice VLAN enabled	Data VLAN enabled	Both Voice and Data VLAN enabled
Network Port (Port 0)	N/A	N/A	N/A
IP Deskphone Port (SMP)	Receiving the frames with Voice VLAN tagged only. Sending the frames with Voice VLAN tagged.	N/A	Receiving the frames with Voice VLAN tagged only. Sending the frames with Voice VLAN tagged.
PC Port (Port 1)	N/A	Tagging the incoming frame untagged and	Tagging the incoming frame untagged and forwarding it to network port.

Ports	Voice VLAN enabled	Data VLAN enabled	Both Voice and Data VLAN enabled
		forwarding it to network port. Replacing the incoming frame tagged with VLAN other than Data-VLAN and forwarding it to network port. Sending all kinds of frames.	Replacing the incoming frame tagged with VLAN other than Data-VLAN and forwarding it to network port. Sending all kinds of frames.

VLAN configuration can be done either manually or through DHCP. Refer to Device Settings on the IP Deskphone with SIP Software on page 101 for more detail on configuring VLANs.

Chapter 21: SIP messages supported by the IP Deskphone

SIP methods

The table below provides a list of SIP messages supported by the Avaya 1200 Series IP Deskphone.

Table 63: SIP methods

Method	Supported?	Comments
INVITE	Yes	Mid-call re-invites for media changes also supported.
ACK	Yes	
BYE	Yes	
CANCEL	Yes	
OPTIONS	Response only	
INFO	Yes	Optionally used for in-session DTMF signaling, and Avaya Call Server specific NAT detection
PING	Yes	Proxy detection, monitoring and Avaya Call Server specific firewall traversal
REGISTER	Yes	For user registration
REFER	Yes	For transfer
NOTIFY	Yes	
SUBSCRIBE	Yes	
PUBLISH	Yes	For VQMon Publish
PRACK	Yes	No support for PRACK-specific early-media negotiation scenarios
MESSAGE	Yes	
UPDATE	Yes	UPDATE messages received in an early dialog state require reliable provisional responses. If PRACK is disabled, or not used by a local or remote party, some

Method	Supported?	Comments
		UPDATE operations fail as described in RFC3311. The support of UPDATE messages is not a configurable feature.

SIP responses

The following SIP responses are also supported:

- 1xx Response—Information Responses
- 2xx Responses—Successful Responses
- 3xx Response—Request Failure Responses
- 4xx Response—Server Failure Responses
- 6xx Response—Global Responses

1xx Response—Information Responses

1xx Response	Send	Receive	Comments
100 Trying	Yes	Yes	The IP Deskphone can generate this response for an incoming INVITE if it has taken too long to generate a 180 response. Upon receiving this response, the IP Deskphone waits for a 180 Ringing, 183 Session Progress, or 200 OK responses.
180 Ringing	Yes	Yes	The IP Deskphone begins local ringing through the active transducer.
181 Call is being forwarded	No	Yes	See 183.
182 Queued	No	Yes	See 183.
183 Session progress	No	Yes	The IP Deskphone accepts a 183 response with SDP to allow for early-media negotiation.

2xx Response—Successful responses

2xx Response	Send	Receive	Comments
200 OK	Yes	Yes	
202 Accepted	Yes	Yes	

3xx Response—Redirection responses

3xx Response	Send	Receive	Comments
300 Multiple Choices	No	Yes	When receiving this response, the IP Deskphone redirects the original request to next contact specified.
301 Moved permanently	No	Yes	When receiving this response, the IP Deskphone redirects the original request to the new contact specified. However, the IP Deskphone takes no additional special consideration of the "permanent" status of this change.
302 Moved temporarily	Yes	Yes	This response is sent to an incoming invite if the IP Deskphone has local call-forwarding enabled. When receiving this response, the IP Deskphone redirects the original request to the new contact specified.
305 Use Proxy	Yes	Yes	The IP Deskphone generates these responses when receiving requests that did not come through the configured SIP proxy. When receiving this request, the IP Deskphone contacts the new address in the Contact header field.
380 Alternate service	No	Yes	When receiving this request the IP Deskphone contacts the new

3xx Response	Send	Receive	Comments
			address in the Contact header field.

4xx Response—Request failure responses

4xx Response	Send	Receive	Comments
400 Bad request	Yes	Yes	The IP Deskphone generates a 400 Bad Request response for various failure conditions generally when a request is invalid, and a more specific error response does not apply.
401 Unauthorized	No	Yes	Receiving a 401 response results in the IP Deskphone re-issuing the request using HTTP digest authentication.
402 Payment required	No	Yes	See default handling.
403 Forbidden	No	Yes	See default handling.
404 Not found	Yes	Yes	The IP Deskphone generates this response for requests to unknown users. Receiving this response falls through to the default handling.
405 Method not allowed	Yes	Yes	The IP Deskphone ends this response to a known method if it is received at a time when the IP Deskphone is not prepared to handle or the request is missing necessary information. Receiving this response falls through to the default handling.
406 Not acceptable	Yes	Yes	The IP Deskphone can send this response when receiving a REFER request which has an unsupported URI. Receiving this response falls through to the default handling.
407 Proxy authentication required	No	Yes	See 401.

4xx Response	Send	Receive	Comments
408 Request timeout	No	Yes	See default handling.
410 Gone	No	Yes	See default handling.
413 Request entity too large	No	Yes	See default handling. The IP Deskphone does not automatically retry if a retry-after header is present.
414 Request---URL too long	No	Yes	See default handling.
415 Unsupported Media	Yes	Yes	The IP Deskphone can send this response when an incorrect content-type is detected for a request. Receiving this response falls through to the default handling.
420 Bad Extension	Yes	Yes	The IP Deskphone can respond with a 420 when checking required extensions of incoming requests. When receiving a 420, see default handling. The IP Deskphone does not retry the request.
480 Temporarily unavailable	No	Yes	See default handling.
481 Call leg/ transaction does not exist	Yes	Yes	Incoming requests are matched against existing dialogs. If a request appears to be in-dialog, but does not have an existing dialog, the IP Deskphone responds with a 481. For incoming 481 responses, the default handling is used.
482 Loop detected	Yes	Yes	Default handling is used when this response is received.
483 Too Many Hops	No	Yes	See default handling.
484 Address Incomplete	No	Yes	See default handling.
485 Ambiguous	No	Yes	See default handling. The IP Deskphone does not attempt to retry the request.
486 Busy Here	Yes	Yes	The IP Deskphone can respond with this if the user is on the IP Deskphone, and the IP Deskphone has reached its

4xx Response	Send	Receive	Comments
			maximum number of allowed calls and cannot present the incoming call to the user. When this message is received by the IP Deskphone an error is displayed and a busy tone is played.
487 Request Canceled	Yes	Yes	See default handling.
488 Not Acceptable	Yes	Yes	The response is used by the IP Deskphone when a failed media negotiation occurs.
491 Request Pending	Yes	Yes	The IP Deskphone sends and receive this message in GLARE conditions.

5xx Response—Server failure responses

5xx Response	Send	Receive	Comments
500 Internal Server Error	Yes	Yes	The IP Deskphone can send this response when a request is received but the IP Deskphone software is not in a correct state to handle it. When receiving this message the IP Deskphone displays an error for the user.
501 Not Implemented	No	Yes	See default handling.
502 Bad Gateway	No	Yes	See default handling.
503 Service Unavailable	Yes	Yes	
504 Gateway timeout	No	Yes	See default handling.
505 Version Not Supported	Yes	Yes	

6xx Response—Global responses

6xx Response	Send	Receive	Comments
600 Busy Everywhere	Yes	Yes	The IP Deskphone can send this response when the IGNORE setting is configured to NETWORK, and the user chooses to ignore an incoming call. When received, this response falls through the default handling.
603 Decline	Yes	Yes	The IP Deskphone can send this response when the user declines an incoming call. An optional reason can be supplied.
604 Does Not Exist Anywhere	No	Yes	See default handling.
606 Not Acceptable	No	Yes	See default handling.

Default error handling

All 4xx/5xx/6xx responses (with the exception of 401/407) received by the IP Deskphone when attempting to initiate a call result in the display of an error on the screen, and typically results in fast or regular busy tone.

If a media negotiation fails during dialog setup, the IP Deskphone terminates the dialog.

If an in-dialog failure occurs during media (re)negotiation, the IP Deskphone falls back to previously negotiated media settings. When a failure occurs that makes this impossible, the IP Deskphone attempts to clear the call by terminating the dialog.

SIP header fields

The following table contains the supported SIP headers.

Header field	Supported?
Accept	Yes

Header field	Supported?
Accept-Encoding	Yes
Accept-Language	Yes
Alert-Info	Yes
Allow	Yes
Allow-Events	Yes
Authentication-Info	Yes
Authorization	Yes
Call-Id	Yes
Call-Info	Yes
Contact	Yes
Content-Disposition	Yes
Content-Encoding	Yes
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes
Expires	Yes
Error-Info	Yes
Max-Forwards	Yes
Mime-Version	Yes
Organization	Yes
P-Access-Network-Info	Yes
P-Asserted-Identity	Yes
P-Associated-URI	Yes
P-Called-Party-ID	Yes
P-Charging-Function-Addresses	Yes
P-Charging-Vector	Yes
P-Media-Authorization	Yes
P-Preferred-Identity	Yes
P-Visited-Network-ID	Yes
Path	Yes

Header field	Supported?
Priority	Yes
Privacy	Yes
Proxy-Authenticate	Yes
Proxy-Require	Yes
RAck	Yes
Reason	Yes
Record-Route	Yes
Refer-To	Yes
Referred-By	Yes
Remote-Party-ID	Yes
Replaces	Yes
Reply-To	Yes
Require	Yes
Resource-Priority	Yes
Retry-After	Yes
Route	Yes
RSeq	Yes
Server	Yes
Service-Route	Yes
Subject	Yes
Supported	Yes
Timestamp	Yes
To	Yes
Unsupported	Yes
User-Agent	Yes
Via	Yes
Warning	Yes
WWW-Authenticate	Yes

Session description protocol usage

SDP Headers	Supported?
v--Protocol version	Yes
o--Owner or creator and session identifier	Yes
s--Session name	Yes
t--Time description	Yes
c--Connection information	Yes
m--Media name and transport address	Yes
a--Media attribute lines	Yes

SDP and Call Hold

The IP Deskphone can support sending and receiving of hold using the method specified by RFC2543 and RFC3261/3264.

Transport layer protocols

Protocol	Supported?
Unicast UDP	Yes
Multicast UDP	No
TCP	No

SIP security authentication

Authentication	Supported?	Comments
Digest Authentication	Yes	
Proxy-to-User Authentication	Yes	
User-to-User Authentication	No	The IP Deskphone responds to a 401, but never challenges incoming requests with a 401 response.
S/MIME	No	
AKA	No	

SIP DTMF Digit transport

Transport type	Supported?
RFC2833	Yes
In-band tones	Yes
Out-of-band tones	Yes (vnd.avaya.digits)

Supported subscriptions

Subscription type	Supported	Avaya Call Server specific
address-book	Yes	Yes
call-park	Yes	Yes
dialog	Yes	Yes
presence	Yes	Yes
message-summary	Yes	No

Subscription type	Supported	Avaya Call Server specific
ua-profile	Partial	Yes
service-package	Yes	Yes
network-redirection-reminder	Yes	Yes

Supported instant messaging

Message type	Supported?
plain text	Yes
Avaya unencrypted	Yes
Avaya encrypted	Yes

Chapter 22: Audio codecs

Overview

The optional audio codecs feature allows you to select the audio compression or decompression algorithm (codec) used on the Avaya IP Deskphone. You provision codecs using the Device Configuration file, and then the user can select from the provisioned codecs using the Audio menu on the Avaya 1200 Series IP Deskphone.

When the user selects an audio codec, that codec is used for both incoming and outgoing calls.

The following table lists the audio codecs supported by IP Deskphone.

Table 64: Audio codecs supported by IP Deskphone

Codecs	Description
G.723.1	This codec is a compressed, nonwideband audio codec. It provides high-quality audio with less network connection requirements. This codec is ideal for bandwidth-conscious environments that do not support higher quality encoding. Expanded support of the existing G.729a codec with Annex allows for two byte Silence Insertion Descriptor (SID) frame for CNG.
G.711 a-law	PCMA
G.711 mu-law	PCMA
G.729	

In the case of an upgrade from a UNIStim IP Deskphone or an earlier version of the SIP firmware, Avaya recommends that you specify the preferred codec in the Device Configuration file; otherwise the default value is used.

The G.711 codec (PCMU and PCMA) is always used to place the codec list for emergency 911 calls. The G.711 codec is always used to receive incoming calls from the emergency operator. If the administrator disables this codec, the SIP IP Deskphone can make outgoing non emergency calls.

You can configure a maximum of 15 codecs. You can enable or disable the use of specific codecs for incoming and outgoing calls, though incoming and outgoing calls are not specifically independent.

VQMON Codec configuration

You can enable the VQMON feature through the Device Configuration file to send an SIP Publish message with the VQMON report as text content, and use this report for QoS monitoring.

Network layer for the SDP negotiations

The following table contains static payload types and other parameters for the supported codecs.

Table 65: Static payload types and other parameters for for the supported codecs for the IP Deskphone

Codec	Payload type	SDP encoding name	Clock rate (HZ)	Bit rate (kbps)	M5T name	ptime (milisec)	Channels
G.711 a-law	8	PCMA	8000		ePCMA	20	1
G.711 u-law	0	PCMU	8000		ePCMU	20	1
G.729A + 40ms ptime	18	G729	8000		eG729	20	1
G.729B	18		8000	8	eG729	20	1
G.723.1	4	G723	8000	5.3 6.3	eG723	30	1
G.723.1A	4		8000	5.3 6.3	eG723	30	1

The annexes selection for G.729 and G.723.1 are not available to the user, and the administrator is responsible for enabling or disabling annexes using Device Configuration flags.

Codec preference through Device Configuration

Use the Device Configuration file to specify a list of codecs, and the preferred order in which they are used for incoming and outgoing calls. You can also use the Device Configuration file to enable or disable AnnexB support by G.729 and AnnexA support by G.723.1. You can add a text descriptor to the technical name of the audio codec; these descriptors appear on the user interface of the IP Deskphone.

You can specify, by name, the exact codecs to offer in the Device Configuration file. This grants the administrator full control over the audio settings used for inbound and outbound calls. The following table is a sample of Device Configuration file entries for audio codec configuration.

Table 66: Sample Device Configuration entries

```
AUDIO_CODEC# CodecID text_description
AUDIO_CODEC1 PCMA standard a-law
AUDIO_CODEC2 PCMU standard u-law
AUDIO_CODEC3 G729 729 codec
AUDIO_CODEC7 G723 high-compression codec
```

The following table lists the codec identifiers for the Device Configuration file.

Table 67: Codec identifiers for the Device Configuration file

PCMA
PCMU
G729
G723

The IP Deskphone displays the codecs listed in the exact order that they are listed in the Device Configuration file.

The list of codecs specified in the Device Configuration file determines the list of codecs that are available for selection on the IP Deskphone.

Two fields in the device configuration file, G729_ENABLE_ANNEXB and G723_ENABLE_ANNEXA are used to enable or disable AnnexB and AnnexA support by G.729 and G.723 codec, respectively. These flags can have the following values: YES, NO (No is the default value).

Important:

If codecs are not specified, the default list used by the current version of the IP Deskphone is PCMU, PCMA, G.729.

To stop the IP Deskphone from using a specific codec, you must change the its entry in the Device Configuration file to a different codec, and then clear the value, which disables the codec entry. If you remove all codecs from the allowed list, the IP Deskphone resets to the default list of codecs.

Important:

To reset the phone to the default list of codecs, it is necessary to remove the values against each AUDIO_CODECx item in the Device Configuration file.

For example:

```
AUDIO_CODEC1 PCMA standard a-law
```

```
AUDIO_CODEC2 PCMU standard u-law
```

```
AUDIO_CODEC3 G729 729 codec
```

```
AUDIO_CODEC4 G722 wideband codec
```

```
AUDIO_CODEC5 G723 high-compression codec
```

would become

```
AUDIO_CODEC1
```

```
AUDIO_CODEC2
```

```
AUDIO_CODEC3
```

```
AUDIO_CODEC4
```

```
AUDIO_CODEC5
```

If the ordered list of codecs is small and no matching codec is found during negotiations, the call drops, as the audio stream cannot be established. For backward compatibility with SIP Firmware Release 1.X, the Device Configuration file supports the **DEF_AUDIO_QUALITY** parameter as long as no codec is allowed using the new parameter **AUDIO_CODECN**, in which case the **DEF_AUDIO_QUALITY** parameter is ignored and has no effect.

Specifying the **DEF_AUDIO_QUALITY** as High or Medium has the same effect as omitting the parameter all together and without specifying codec through the new parameters.

If set to Low, then the list of default codecs is reversed before being sent in the SDP negotiations. When you do not provide a text description in the Device Configuration file, the application uses the default text description from the language file.

The **AUDIO_CODECN** parameters specifies the order of preference for audio codecs. If there are no valid entries provided, then the parameter uses the default list of codecs. If you enter a codec that is not recognized by the IP Deskphone then the parameter considers the codec as a blank entry. To remove a codec from the list, you must first blank the entry, or change it to an invalid codec name in the Device Configuration file.

Codec preference selection on the IP Deskphone

The Audio Quality Settings screen on the IP Deskphone allows the user to select an exact codec by name. This grants the user full control over the audio settings used for inbound and outbound calls.

The list of codecs is populated with the names of the codecs provided during Device Configuration. If a text descriptor is provided for a codec in the Device Configuration file, it appears after the codec name. The Audio Codec Ordering screen allows the user to modify the order of preference of the codecs. To change the list of available codecs, you must perform an update through Device Configuration. The IP Deskphone creates the ordered list from the list of codecs in the Device Configuration file. The user can reorder the list using the

Preferences menu. On subsequent Device Configuration updates, at start time, or other updates, the ordered codec list of the user is synchronized with the list in the Device Configuration file. This synchronization makes both lists equal. If the user creates an order that is different from the one in the Device Configuration file, the IP Deskphone appends it to the end of the list.

Codecs preferences on the IP Deskphone

The user cannot modify the text descriptors through the IP Deskphone; the text descriptors can only be read by the user. After the system loads the Device Configuration file, the user preference selections are synchronized with the system codecs specified in the Device Configuration file. This ensures that the codecs available to the user are always set according to user preferences.

If the user modifies the order through the IP Deskphone, then the user-defined order is saved for the codecs that are defined as system codecs in the Device Configuration file. Codecs are appended at the end of the list in their relative order from the Device Configuration file. Until the user modifies the order of the codecs, the list of ordered codecs reflects the order specified in the Device Configuration file.

The following table shows examples of the list of codecs provided by Device Configuration, user configuration, and resulting list of codecs that the system uses for presentation and codec negotiation purposes.

Table 68: Examples of the ordered lists of Codecs

Supported by IP-set	Ordered list of codecs provided by Device Configuration	Ordered list of codecs provided by user configuration	Ordered list of codecs used by the SIP IP Deskphone
	A, B, C, D, E	N/A	A, B, C, D, E
	A, B, C, D, E	E, D, C, B, A	E, D, C, B, A
A, B, C, D, E, F, G	A, B, C, D, E	A, D, E	A, D, E, B, C
	A, C, D, E	A, B, C, D, E	A, C, D, E
	A, C, D, E	A, B, C, E	A, C, E, D

Chapter 23: Certificate-based authentication

Feature Overview

Certificate-based authentication allows the administrator to ensure that the IP Deskphone is authorized to access the enterprise LAN environment. Certificate-based authentication supports three types of Extensible Authentication Protocols (EAP):

- EAP-MD5—User ID/password-based authentication
- EAP-PEAP—certificate-based authentication
- EAP-TLS—certificate-based authentication

Trusted root certificates and device certificates must be installed before using EAP-TLS, EAP-PEAP or HTTPS.

Certificate-based authentication supports two types of device certificates: one is used by EAP-TLA, and the other is used by SIP-TLS, but the administrator can also have a third device certificate for HTTPS. The user must connect to a Certificate Authority (CA) to retrieve or sign certificates. A CA is a trusted third party; components of a system agree to trust the CA to verify the necessary information.

When the CA validates the user information, it issues the user a certificate that contains a variety of data, including:

- the identity of the issuing CA
- how much the CA trusts the user
- an expiry date for the certificate

Other components of the system can read the user's certificate to determine if the certificate, and the identity it represents, are valid.

The administrator can install and manage the certificates on the IP Deskphone. The certificates authenticate the IP Deskphone to an authentication server before the IP Deskphone can access the enterprise network.

Certificate-based authentication includes the following features:

- EAP Authentication

The supplicant can be authenticated to an authentication server using one of these EAP methods:

- EAP-MD5
- EAP-PEAP
- EAP-TLS

- Device certificate management

  The administrator can install a device certificate on the IP Deskphone by using SCEP or PKCS#12 import file. The IP Deskphone can verify the imported device certificate by checking the availability of the IP Deskphone against the Certificate Trust List (CTL) stored in the IP Deskphone. CTL is a predefined list of trusted certificates including CAs, intermediate CAs, and server certificates which the IP Deskphone views as trust anchors. The administrator can also view and delete a certificate on the IP Deskphone.

- Provisioning configuration File

  The provisioning configuration file, such as 12xxSIP.cfg and all other configuration files referred by 12xxSIP.cfg, specifies software and resource files that are downloaded to the IP Deskphone from a provisioning server by using the secure provision method HTTPS.

- Security and error logs

  The administrator can view security and error logs that occurred during the operation of the IP Deskphone. This feature is accessed through the Diagnostics screen.

- Security policy file updates

  The security policy file defines a set of rules to determine the required actions taken by the IP Deskphone.

Certificates overview

Certificates bind an identity to a pair of electronic keys that are used to encrypt and sign digital information, and make it possible to verify someone's claim that they have the right to use a given key. Certificates provide a complete security solution, assuring the identity of all parties involved in a transaction. Certificates are issued by a Certification Authority (CA) and are signed with the CA's private key.

A certificate contains the following information:

- Owner's public key
- Owner's name
- Expiration date of the public key
- Name of the issuer (the CA that issued the certificate)

- Serial number of the certificate
- Digital signature of the issuer

Root certificate installation

The customer root certificate is a self-signed certificate (a self-issued certificate where the subject and issue fields contain identical DNs, and are not empty). The customer root certificate must be installed on the IP Deskphone and stored in the IP Deskphone trusted store for the following reasons:

- to verify the identity of the various servers that the IP Deskphone may attempt to establish secure connections with (such as TLS and HTTPS)
- to authenticate the signatures on software and configuration files that you download onto the IP Deskphone.

You can install a customer root certificate by using Simple Certificate Enrollment Protocol (SCEP) or by using the configuration file (for example 12xxSIP.cfg.).

If you use SCEP, you must first configure the URL of the CA SCEP server and the domain name, and then you can connect to the CA and download a CA root certificate to the IP Deskphone.

- The IP Deskphone sends the GetCACert request to the SCEP-enabled interface for a CA server.
- The IP Deskphone waits for a response. If an error is received (such as timeout or server unreachable), the registration process ends.
- The IP Deskphone accepts the reply which contains the CA root certificate. The reply may also include one or two Registration Authority (RA) certificates which are stored temporarily for use during the request for a device certificate.
- If the CA root certificate is not already on the IP Deskphone, the fingerprint is computed and displayed. The computed fingerprint is the thumbprint of the certificate (the SHA1 hash of the public key of the certificate).
- You must Accept or Reject the fingerprint.
- If the CA root certificate is rejected, the registration process ends.
- If the CA root certificate is already in the trusted store, no prompt appears.
- If the fingerprint is accepted, the CA root certificate is added to the trusted store on the IP Deskphone.

If you use the configuration file (for example, 11xxe.cfg), you can download one or more CA root certificates to the IP Deskphone.

- The [USER_KEYS] section is added to the configuration file (for example 12xxSIP.cfg), where the FILENAME attribute points to the file name of a customer root certificate in Privacy Enhanced Mail (PEM) format. The PROTOCOL attribute of the [USER_KEYS] section can be assigned to one of the IP Deskphone supported protocols, such as HTTP, TFTP, HTTPS and FTP.

- After the configuration file is downloaded and parsed by the IP Deskphone, the [USER_KEYS] section is processed and the root certificate is downloaded to the IP Deskphone.
- After the certificate file is downloaded, you must authenticate the contents of the certificate file before installing it on the IP Deskphone. There are two possible situations.
    - If there are no existing customer root certificates on the IP Deskphone, a fingerprint for the file is computed. Depending on the value that is configured in the Security Policy parameter, CUST_CERT_ACCEPT, the user can either be prompted to accept this fingerprint, or prompted to enter the fingerprint for verification.
    - If there is one or more customer root certificate on the IP Deskphone, the certificate file must be digitally signed with a signing certificate. In this case, there is no interaction with the user. The signature is internally verified and the signing certificate is verified to be issued by a customer root certificate that is already installed on the IP Deskphone.
- If the authentication of the file is successful, the customer root certificate is installed on the IP Deskphone in the trusted certificate store.

Important:

Although the certificate file usually contains a single customer root certificate, it is possible that the certificate file may contain more than one certificate and CRL. This occurs where the PEM encoding for each certificate or CRL is appended in the file with a blank line between each file. If the authenticity of the file is successfully verified, all entities in the file are installed on the IP Deskphone.

Signing a resource file

The following is the command to sign a resource file using `openssl` .

```
openssl smime –sign –in unsigned_file –signer sign_cert_file –outform
PEM –binary –inkey sign_cert_pk_file –out tmp_signature_file
```

The first customer root certificate must either be signed by a Avaya Trusted Certificate or Fingerprint accepted. To control further signing of a customer root certificate, and prevent security risks, the following Security Policy parameter must be configured.

```
CUST_CERT_ACCEPT — VAL_NO_CHECK
```

When the IP Deskphone tries to establish a secure connection (for example, HTTPS, SIP TLS) with a server, the server provides its certificate which then must be verified by the IP Deskphone.

The following are the possible configurations (depending on the server configuration):

1. Server can provide only its Server certificate.
2. Server can provide the entire certificate chain (up to the Root CA certificate).

In the first scenario, the IP Deskphone only needs the CA certificate which was used to sign the Server certificate. The certificate file must be PEM encoded.

In the second scenario, every certificate in the chain must be verified. Root and Intermediate CA certificates of the chain must be installed in the IP Deskphone Trusted Certificates store. Certificates must be PEM encoded and combined into one file.

Device certificate installation

A device certificate is a certificate used to prove the identity of the IP Deskphone to a server while establishing various secure connections, such as TLS and HTTPS, between the IP Deskphone and a server. Each device certificate is associated with a specific usage purpose. It is possible for one or two device certificates to be installed on the IP Deskphone (for example, one for all TLS connections and one for VPN). A Device Certificate Profile (DCP) allows for various combinations of sharing device certificates among different applications. Within the DCP, you can identify one of more uses (or purposes) for the device certificate associated with each profile, to provide a flexible model for the sharing of device certificates among IP Deskphone applications.

The following sections describe the process used to install a device certificate on the IP Deskphone. This process starts with defining a DCP for each device certificate that must be installed on the IP Deskphone. See Device certificate profiles on page 202.

The two methods used to install a device certificate on the IP Deskphone are:

- SCEP
- PKCS#12 download

SCEP is a protocol that allows the IP Deskphone to send a device certificate request to a CA server based on a locally generated private key to provide more security for the private key (because the private key is never transmitted, even in an encrypted form). See SCEP on page 207

PKCS#12 is an industry standard for exchanging certificate and private keys. A device certificatd downloaded to the IP Deskphone in a PKCS#12 file contains the complete certificate including the private key of the device certificate which is generated offline by a Certificate Authority (CA). The PKCS#12 file is encrypted using password at the time of generation to protect the private key. See PKCS 12 download on page 209.

For more information on defining a device certificate profile, see Device certificate profiles on page 202.

Device certificate profiles

You can determine the method used to install a device certificate on the IP Deskphone. Each device certificate installed on the IP Deskphone is attached to a Device Certificate Profile (DCP). The configuration of the profiles allows you to determine the method used to install a device certificate and provides you with some control over the device certificate attributes.

You can do the following:

- Specify the method used to obtain a device certificate for the IP Deskphone (SCEP or PKCS#12).
- Specify the purpose of a device certificate; whether the certificate is used for EAP-TLS, or HTTPS (for example, allow sharing of device certificates).
- Renew a device certificate obtained by SCEP.
- Customize attributes requested from a SCEP server such as the Distinguished Name (DN).

The following table defines the profile attributes and the allowed values for a device certificate profile.

Table 69: Device Certificate Profile Attribute

Name	Type	Value(s)	Default	PKCS#12 Required	Description
Index	Int	>0	Pre-defined [1-MAX_PROFILES]	✔	Device Certificate Profile index.
Version	String		""	✔	String containing version of last installed PKCS12 file.
Source	Int	0 = SCEP 1 = PKCS12	Index 1 = 0 Index 2+ = 1	✔	SCEP is default for the first profile. PKCS12 is default for the other profiles.
Active	Int	0 = Inactive 1 = Device 2 = User (future)	Index 1 = 0 Index 2+ = 1	✔	Specifies if the profile is active and if the profile is used for device or user authentication. The value 0 indicates that the device certificate

Name	Type	Value(s)	Default	PKCS#12 Required	Description
					with this index is not used (regardless of the Source value).
Purpose	Int	Bit flags	–1 (ALL)	✔	Covers all feature usages plus special cases for All(=–1).
Delete	Int	0 = No 1 = Yes	0	✔	Used to force a device certificate to be deleted. Automatically resets to 0 after a certificate is deleted.
CAServerName	String		AdminCA1	✘	AdminCA1 is a default for backward compatibility with previous UNIStim versions.
HostnameOverride	String		""	✘	Override hostname for this certificate only (only for SCEP). The default is empty because the default is not used.
Renew	Int	-1 = Never 0 = Immediate >0 = # Days	30	✘	Number of days remaining to request a new device certificate.
AutoCN	Boo1	0 = Manual 1 = Auto	1	✘	Auto means that the common name (CN) is automatically populated with the UPN as in UNIStim 3 (for example: hostname@domainname). This is provided for

Name	Type	Value(s)	Default	PKCS#12 Required	Description
					backward compatibility.
CN	String		""	✘	Common Name
O	String		""	✘	Organization
OU	String		""	✘	Organizational Unit
S	String		""	✘	Province/State
C	String		""	✘	Country
Key Usage	Int		0x00a0	✘	For example: Digital Signature + Key Encipherment. Default is TLS compatible.
Extended Key Usage	Int		2 (clientAuth)	✘	For example: clientAuth. Default is TLS client compatible.
SubjAltName				✘	Following are Subject Alternative Name fields that must be specified.
FQDN	Boo1		0	✘	Include in SCEP request if configured. Content from current hostname and domain name configurations.
USER_FQDN	Boo1		0	✘	Include in SCEP request if configured. Content from current hostname and domain name configurations.
IPAddress	Boo1		0	✘	Include in SCEP request if configured.

Name	Type	Value(s)	Default	PKCS#12 Required	Description
					Content from current hostname and domain name configurations.

With the exception of the version, all the DCP configurations described in the preceding table apply to SCEP-requested device certificates. The PKCS#12 column identifies the limited set of parameter that apply to a DCP configured for PKCS#12. Many of the parameters apply only to the configuration of a certificate request, which is why the parameters apply only to SCEP.

The following describes the key profile attributes:

- Index The Index is the index of the device certificate profile. For each type of IP Deskphone, there is a fixed number of profiles available in the range of 1 to MAX_PROFILES. The index also identifies a priority. When a device certificate is requested for a specific purpose, such as EAP-TLS, the IP Deskphone searches through the device certificates to find the first one that is defined, active and can be used for the requested purpose.
- Source The Source identifies if the IP Deskphone requests the device certificate using SCEP or if the device certificate is downloaded using PKCS#12. If PKCS#12 is specified, direct action is not taken. This allows a downloaded device certificate to be installed in this profile.
- Active If the Active attribute is not active, the IP Deskphone assumes that there is no device certificate associated with the profile and takes no action to request one (even if SCEP is specified as the source).
- CAServerName CAServerName is the name of a CA server that is sent in the initial SCEP request to get the CA root certificate. Although some SCEP servers ignore the CAServerName, the CAServerName is important for EJBCA, and to differentiate between multiple CAs on a single server.

  Important:

  CAServerName must not be confused with the URL specified for the CA server which is used to make the SCEP connection.
- AutoCN The AutoCN parameter indicates if the CN in an SCEP certificate request should be automatically populated based on the Hostname and Domain Name configuration parameters. The Hostname and Domain Name parameters are part of the overall IP Deskphone configuration and are not configurable within each DCP.
  - If AutoCN is configured as 1 (True), then the CN is constructed as Hostname@Domainname.
  - If AutoCN is configured as 0 (False), then the CN is configured as the value of the CN parameter in the DCP.
- Purpose The Purpose attribute uses bit masks to identify what features a particular device certificate is used for. Two bytes allows for any combination of up to 16 uses. For example,

a certificate that is used for EAP-TLS, DTLS and SCR have the purpose value of 97 (1+32+64).

The following table defines the values of the device certificate profile purposes.

Table 70: Device certificate profile purpose definitions

Application purpose of usage	Value (hexidecimal)	Value
EAP-TLS	0x0001	1
SIP-TLS	0x0002	2
HTTPS	0x0004	4
LICENSING	0x0080	128
ALL	0xffff	-1

The default configurations for DCP #1 allow DCP #1 to be active and to use SCEP to retrieve a device certificate that can be shared among all applications (purpose is ALL). All remaining profiles are configured for PKCS#12 and to be inactive by default. The default configurations are compatible with UNIStim 4 software. SIP 3.0 supports two profiles; SCEP and PKCS12.

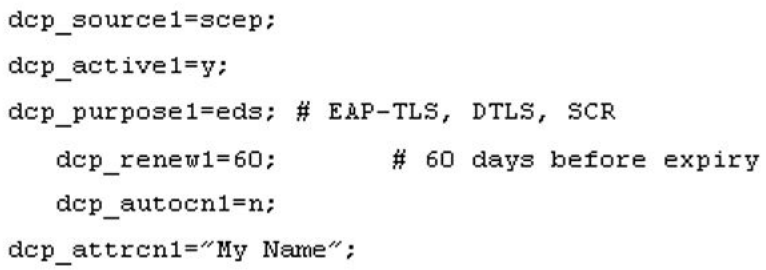
To configure applications on the IP Deskphone, you must know which certificates are required and what methods of device certificate installation are available. You can use this knowledge to determine which profiles must be configured and how the certificates are shared among the different applications. You must als

To configure applications on the IP Deskphone, you must know the following information:

- the required certificates
- the methods of device certificate installation
- the profiles that must be configured
- the method of sharing certificates among different applications
- the certificate attribute requirements (such as, subject, subjAltName, and key usage) for each use

The profile index is part of the provisioning parameter name. For example, the parameter to assign the source (SCEP or PKCS#12) for DCP #2 is the following: `dcpsource2`

The following is an example of the provisioning file, system.prv, that shows some of the device certificate profile attributes that are provisioned when SCEP is used to install a device certificate.



```
dcp_source1=scep;
dcp_active1=y;
dcp_purpose1=eds; # EAP-TLS, DTLS, SCR
   dcp_renew1=60;       # 60 days before expiry
   dcp_autocn1=n;
dcp_attrcn1="My Name";
```

Figure 26: Example of the provisioning file, system.prv

SCEP

Simple Certificate Enrollment Protocol (SCEP) is a process used to obtain a certificate. This process occurs between the IP Deskphone that requires a certificate and a trusted CA that is responsible for providing certificates.

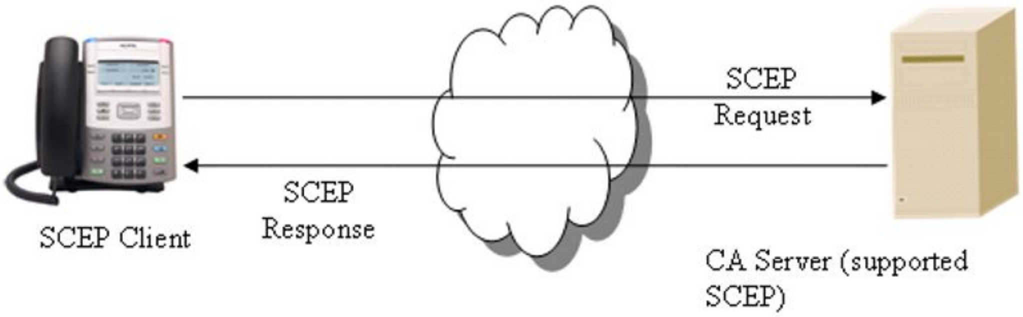


Figure 27: SCEP Client-Server interaction

The IP Deskphone can require several device certificates. You can request an individual device certificate for each application, or you can request a device certificate to be shared among applications.

The following describes the enrollment process for the IP Deskphone for which a device certificate profile is properly configured for SCEP. The following process is executed by the IP Deskphone for every active Device Certificate Profile (DCP) on the IP Deskphone that is configured for SCEP.

1. After the IP Deskphone starts up, the IP Deskphone automatically generates a private-public key pair for each Device Certificate Profile configured on the IP Deskphone for SCEP.
2. The IP Deskphone uses the SCEP GetCACert command to retrieve a customer root certificate from the CA server and prompts the administrator to validate the

certificate fingerprint before the IP Deskphone stores the root certificate permanently on the IP Deskphone.

3. The IP Deskphone prompts the user to enter a password to be included in the certificate request the IP Deskphone is about to generate. A password may or may not be required depending on the configuration of the SCEP/CA server.
4. The IP Deskphone generates a device certificate request which is forwarded to the certificate authority using the SCEP command PKCSReq.
5. After the device certificate request is approved, the CA signs the device certificate request with the CA private key and returns the completed certificate to the IP Deskphone.
6. The IP Deskphone stores the device certificate and the IP Deskphone private key into the IP Deskphone memory with the matching private key.
7. The IP Deskphone can now verify the identity of the device certificate when requested by a server.

During the enrollment process, and before the IP Deskphone sends the device certificate request to the CA server, the IP Deskphone prompts the administrator to enter a challenge password. The use of a password is optional depending on the configuration of the SCEP server. If the SCEP server is configured to not require a password, the administrator does not enter a value and presses the OK Context-sensitive soft key.

The name included in the device certificate request is constructed using the hostname and domain name shown in the Network Configuration screen immediately under the CA server. If there is no hostname entered, a hostname is created using the IP Deskphone MAC address according to the form NTIPP012345, where NTIPP is an acronym for IP Deskphone and 012345 are the last six hex digits of the MAC address. By default, the certificate request includes a Subject Common Name in the form of hostname@domainname. The SCEP configuration fields in each DCP provide more flexibility in the form and location of this name.

Device Certificate Authentication Considerations for SCEP

An important aspect of the device certificate request is the format and location of the name that is requested for the device certificate. The server presented with a device certificate by the IP Deskphone always confirms the authenticity of the certificate by verifying that the issuer of the device certificate is trusted by the server and that the signature on the device certificate is authentic by performing certificate chain validation. A server also performs verification based on the name contained in the device certificate. Therefore, the name contained in the device certificate must be appropriate to the type of authentication that the server uses. The Subject Common Name (CN), the full Subject Distinguised Name (DN), or the Subject Alternate Name (SAN) is used to determine if the entity has the necessary permissions.

For example, if Microsoft IAS is used as the RADIUS server for EAP-TLS authentication, the CN in the certificate must be the User Principle Name (UPN) of a valid user registered in the

Active Directory configured for remote access. Other RADIUS or TLS servers can impose different conditions on the certificate name.

Important:

Before deploying any solution, you must identify what certificate validation criteria is enforced so that the correct certificate name is requested by the IP Deskphone.

Important:

Some SCEP servers reject all SCEP certificate signing requests that include a Subject Alternate Name (SAN). The Microsoft Windows 2003 Server version of SCEP is an example where a certificate request which includes a SAN is always rejected.

During the enrollment process and before the IP Deskphone sends the device certificate request to the CA server, the IP Deskphone prompts you to enter a challenge password. If the password feature is disabled in the SCEP server, you do not require a password.

A certificate requested by SCEP is stored in Profile 0 and uses some hard-coded attributes for requested certificates.

The following table lists additional provisioning file parameters for SCEP support in addition to UI parameters in the Device settings window.

Table 71: SCEP provisioning parameters

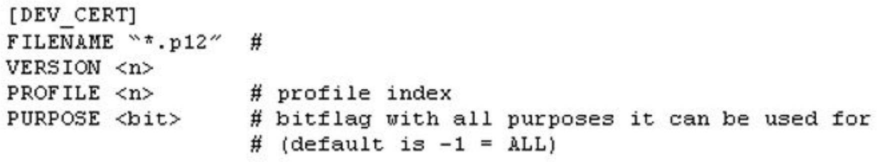
Parameter	Purpose	Default	Allowed
CA	SCEP server	Empty	String Example: http://47.11.15.206/certsrv/mscep/mscep-err.dll
CA_DOMAIN	Domain information used in SCEP request	Empty	String Example: IpClients.com
HOST_NAME	Host name information used in SCEP request	Empty	String Example: 1234

PKCS 12 download

PKCS#12 is an industry standard for importing and exporting keys and their related certificates. On the IP Deskphone, this method is only used to import the IP Deskphone device certificate and private key.

The IP Deskphone can download a PKCS#12 file from the provisioning server. The provisioning configuration file (for example, 11xxe.cfg), contains the [DEV_CERT] section where the FILENAME attribute points to the PKCS#12 file name. The file name must include the * symbol which is substituted with the IP Deskphone MAC address to allow the definition of unique filenames for the PKCS#12 files containing the device certificates for each IP Deskphone.

The following is an example of the [DEV_CERT] section:



```
[DEV_CERT]
FILENAME “*.p12”  #
VERSION <n>
PROFILE <n>       # profile index
PURPOSE <bit>     # bitflag with all purposes it can be used for
                  # (default is -1 = ALL)
```

Figure 28: Example of the [DEV_CERT] section

The administrator is responsible for creating the PKCS#12 file with the required device certificate associated with the private key of the device certificate. The PKCS#12 file must be in Distinguished Encoding Rules (DER) or BER format. If you are creating the certificate for the first time, you must mark the private key of the certificate as exportable. If you export a certificate to a PKCS#12 file, you must enter a password.

Important:

The PKCS#12 password cannot exceed 12 characters in length and must include only characters that you can enter on the IP Deskphone. These characters include all numbers, upper and lower case letters, and the following special characters: _ - . ! @ $ % & + : ^

Installing a device certificate using PKCS 12

The high level sequence of procedures for installing a device certificate using a PKCS#12 file is as follows:

1. The PROFILE Index can range from 1 to the maximum number of supported Device Certificate Profiles (DCP) for the IP Deskphone type.

   Configure the DCP for the specified index for a PKCS#12 downloaded certificate, otherwise the file is rejected. By default, profile 1 is configured for SCEP and all other profiles are configured for PKCS#12.
2. The IP Deskphone checks the version in the [DEV_CERT] section against the version stored in the specified PROFILE. If the version in the specified profile is missing or is older, the device certificate file is downloaded. The profile index is 1.
3. Download the file.
4. Enter the PKCS#12 protected password.

5. Validate the device certificate to ensure that you entered the correct password.
6. Extract the private key and device certificate.
7. Validate the device certificate to ensure the following:
   - the correct password is entered
   - Key size is >= to the value specified in the Security Policy File
   - Key Algorithm is DSA
   - the certificate is not revoked
   - the certificate is not expired
8. If the IP Deskphone has correctly validated the device certificate, the IP Deskphone stores the device certificate and private key in the device certificate profile specified in the [DEV_CERT] section of the IP Deskphone memory (SFS).

   The version specified in the [DEV_CERT] section is stored in the profile for future reference when determining if a new device certificate is available for download.

The PKCS#12 imported certificate is stored in Profile 1.

Certificate Trust Line (certificate verification)

There are two methods to validate a certificate before the IP Deskphone can use it:

- Certificate Revocation List (CRL) — The Certificate Revocation List method has a limitation in the number of CRL entries used due to the limitation of the IP Deskphone memory. It supports up to 100 CRL entries.
- Certificate Trust List (CTL) — The Certificate Trust Line is a collection of certificates bundled together into a file and downloaded into the IP Deskphone. The file is signed and all of the certificates in the bundle are inherently trusted by the IP Deskphone (id the file signature is verified). You can use the CTL in place of a CRL because in the IP Deskphone, the CTL is much smaller than the CRL.

The IP Deskphone uses CTL to verify the various network elements such as proxy servers and provisioning servers. For the IP Deskphone to trust any network element, the certificate of the IP Deskphone must be added to the CTL.

The use of CTL is optional. If CTL is not installed on the IP Deskphone, the authentication of the network element reverts back to the default which is to authenticate the certificate chain to a root certificate trusted by the IP Deskphone.

A file is signed by appending a digital signature which is created using a Signing Certificate. The Signing Certificate must either be directly issued by a CA root certificate installed on the IP Deskphone, or there must be a certificate chain that can be followed which ends with a CA root certificate installed on the IP Deskphone. In either case, the IP Deskphone must have a trust anchor which can verify the authenticity of the Signing Certificate.

The file Signing Certificate requires the following minimum attributes:

- Version—3
- Key usage—Digital Signature
- Extended key usage—Code signing and secure email

Key—1024 or 2048 bits

In addition, the Signing Certificate cannot be a self-signed root certificate and must have a valid Subject Key Identifier and an Authority Key Identifier (which uniquely identifies the issuing certificates).

Validating a certificate using the Certificate Trust List

The high level sequence of procedures for validating a certificate using the Certificat Trust List is as follows:

1. Create the CTL file including start date, expire date and a list of certificates concatenated together in PEM format so that the entire file can be signed by a trusted entity. A signed CTL file consists of the following:

   - Validity fields
   - `NOT_VALID_BEFORE: 23/11/2007 11:12:13`
   - `NOT_VALID_AFTER: 25/10/2011: 22:23:24`
   - Original unsigned file content
   - Digital signature

   The parts are appended together with the Validity periods first, followed by the certificates, and then by the digital signature. The signature must be in the form of a PKCS7 detached signature of the file in PEM format. A detached signature is a signature that does not embed the content that is signed.

   The IP Deskphone does not accept unsigned CTL files. After a CTL file is accepted, the included certificates are added to the trusted certificate store of the IP Deskphone.

   Important:

   Do not insert additional characters between the Certificate and the Digital Signature. Otherwise, the validation fails. Do not change any information from the original file content that was used to create the signature. Otherwise the signature becomes invalid and you must create a new signature.

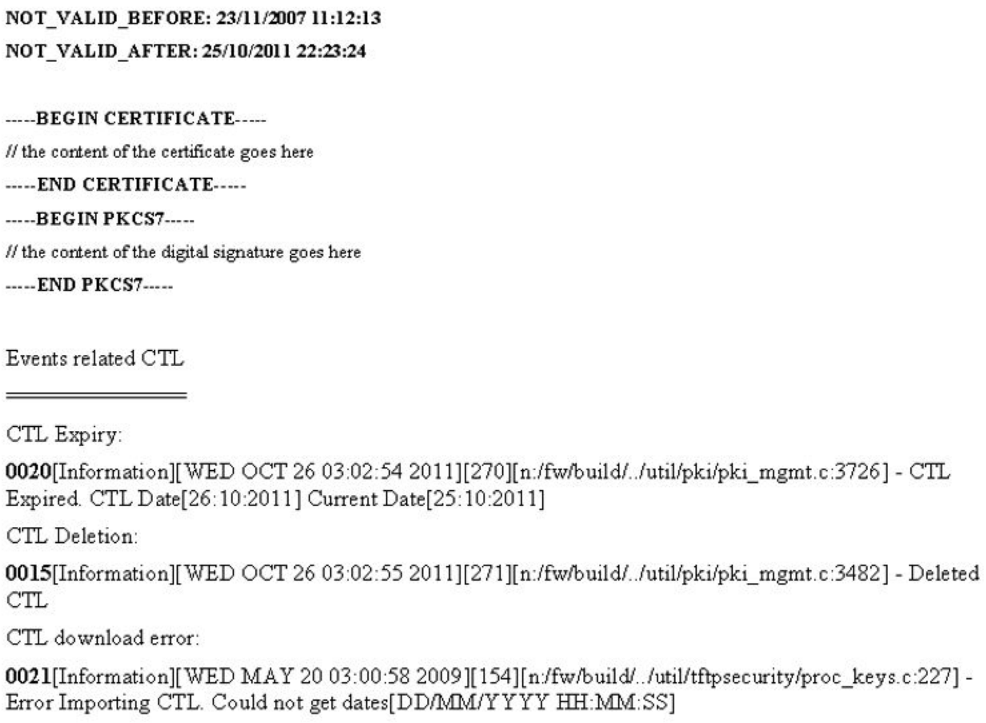
2. The CTL is provisioned to the IP Deskphone in a secure way. Avaya recommends that you use HTTPS as the secure method to download the CTL file to the IP Deskphone.

3. The IP Deskphone checks the validity periods as follows:
    - Not Valid Before—Not used
    - Not Valid After—The IP Deskphone checks this when
        - The CTL file is downloaded.
        - Every 24 hours.
        - When a remote certificate is presented to the IP Deskphone.
        - The CTL is expired; the CTL is deleted and an event is logged in the security log.
4. After the IP Deskphone starts a TLS channel with a server (EAP or TLS) and receives a server certificate, the IP Deskphone validates the certificate by checking the availability of the certificate in the CTL and to decide whether to trust the certificate or not. If the server certificate is not in the CTL, the server certificate is rejected and a TLS channel is not established.

The administrator has to ensure that the CTL is up-to-date. If a new CTL is downloaded to the IP Deskphone, the old CTL file is overwritten by the new one.

The IP Deskphone can trust up to ten server certificates in the CTL file.

The following is an example of a CTL file.



```
NOT_VALID_BEFORE: 23/11/2007 11:12:13
NOT_VALID_AFTER: 25/10/2011 22:23:24

-----BEGIN CERTIFICATE-----
// the content of the certificate goes here
-----END CERTIFICATE-----
-----BEGIN PKCS7-----
// the content of the digital signature goes here
-----END PKCS7-----

Events related CTL
════════════════
CTL Expiry:
0020[Information][WED OCT 26 03:02:54 2011][270][n:/fw/build/../util/pki/pki_mgmt.c:3726] - CTL
Expired. CTL Date[26:10:2011] Current Date[25:10:2011]
CTL Deletion:
0015[Information][WED OCT 26 03:02:55 2011][271][n:/fw/build/../util/pki/pki_mgmt.c:3482] - Deleted
CTL
CTL download error:
0021[Information][WED MAY 20 03:00:58 2009][154][n:/fw/build/../util/tftpsecurity/proc_keys.c:227] -
Error Importing CTL. Could not get dates[DD/MM/YYYY HH:MM:SS]
```

Figure 29: Example of a CTL file

Certificate Administration

The administrator can view and delete certificates. Because a certificate can be deleted, it is critical that the administrator password to access this function is protected and limited only to those individuals who absolutely require it.

Certificate administration is accessed through the Diagnostics menu .

To view Certificate Administration option in Diagnostics menu, configure the following parameter in Security Policy:

```
CERT_ADMIN_UI_ENABLE YES
```

The default value is NO.

After the Security Policy file is enabled, to access the Certificate Administration screen, from the Network screen, choose, Device Settings, Diagnostics, and then Certificate Administration.

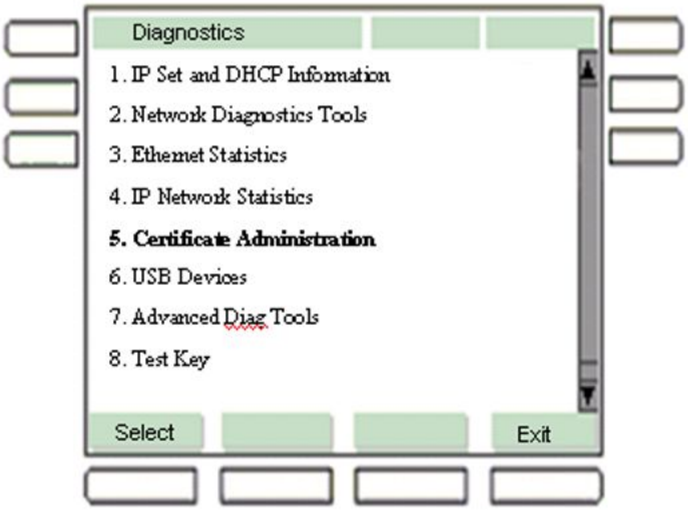


Figure 30: Diagnostics main menu

The following table describes the function of the Navigation keys for the Diagnostics menu.

Table 72: Navigation

Key	Action
Up and down arrows	Use the up and down arrows to change the selected item in the list.
Enter	Invokes the Select Context-sensitive soft key.

Key	Action
Digital keys (number associated with option)	Invokes an appropriate option.
*	Selects the first option Server Settings, but does not activate it.
#	Selects the last option Lock, but does not activate it.

Certificates Administration main menu

The certificates administration screen displays the following options:

- Trusted Certificates
- Device Certificates
- CRL
- CTL

To access the Certificates Administration screen, from the Diagnostics menu, select Certificates Administration.

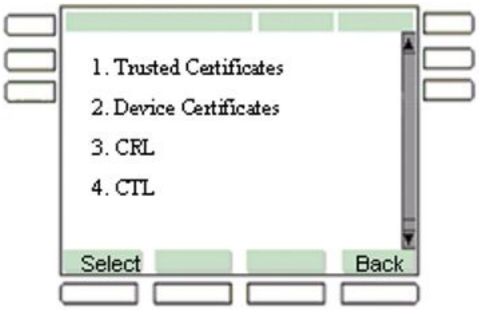


Figure 31: Certificates administration main menu

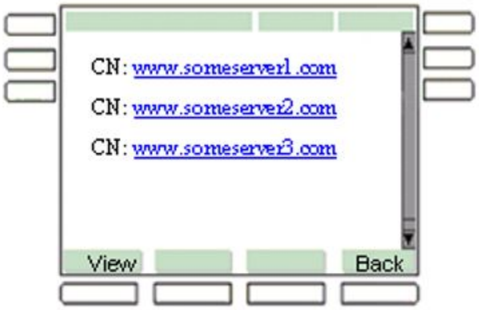
The following table describes the function of the Context-sensitive soft keys for the Certificates Administration screen.

Table 73: Context-sensitive soft keys for the Certificates Administration screen

Context-sensitive soft key	Action
Select	Selects the required option.
Back	Returns you to the Diagnostics menu.

Trusted Certificates screen

The Trusted Certificates screen displays a list of subject Common Name (CN) of the trusted certificates as shown in the following figure:



CN: www.someserver1.com

CN: www.someserver2.com

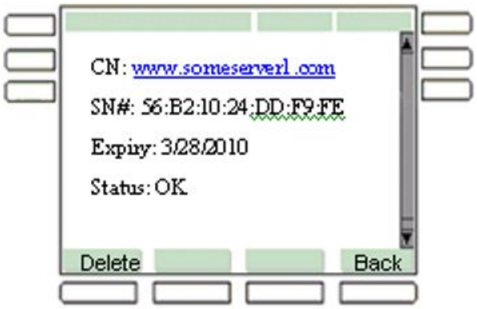
CN: www.someserver3.com

Figure 32: Trusted Certificates screen

The following table describes the function of the Context-sensitive soft keys for the Trusted Certificates screen.

Table 74: Context-sensitive soft keys for the Trusted Certificates screen

Context-sensitive soft key	Action
View	Displays the information of the selected Trusted Certificate which includes the following: • Common Name (CN) • Serial Number (SN#) • Expiry Date • Certificate Status (such as OK or Expired)
Back	Returns you to the previous screen.



CN: www.someserver1.com

SN#: 56:B2:10:24:DD:F9:FE

Expiry: 3/28/2010

Status: OK

Figure 33: Trusted Certificates details

The administrator can delete the certificate in the "Detailed Mode" by using the Delete Context-sensitive soft key. Deletion does not happen automatically; the IP Deskphone displays a warning confirmation screen.

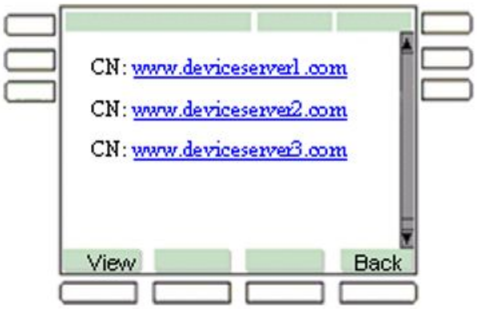
The following table describes the function of the context-sensitive soft keys for the Trusted Certificates Details screen.

Table 75: Context-sensitive soft keys for the Trusted Certificates Details screen

Context-sensitive soft key	Action
Delete	Displays a warning confirmation. Deletes the selected certificate.
Back	Returns you to the previous screen.

Device Certificates screen

The Device Certificates screen displays a list of subject Common Name (CN) of device certificates as shown in the following figure:



CN: www.deviceserver1.com

CN: www.deviceserver2.com

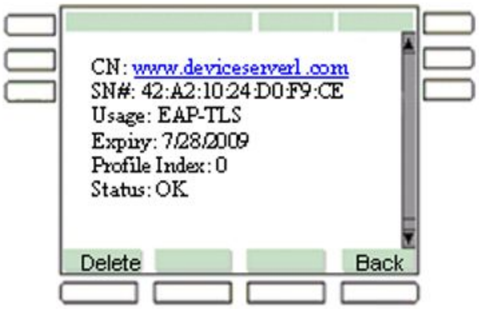
CN: www.deviceserver3.com

Figure 34: Device Certificates screen

The following table describes the function of the Context-sensitive soft keys for the Device Certificates screen.

Table 76: Context-sensitive soft keys for the Device Certificates screen

Context-sensitive soft key	Action
View	Displays the information of the selected Device Certificate which includes the following: • Common Name (CN) • Serial Number (SN#) • Usage • Expiry Date • certificate profile index • Status (such as, OK or Expired)
Back	Returns you to the previous screen.



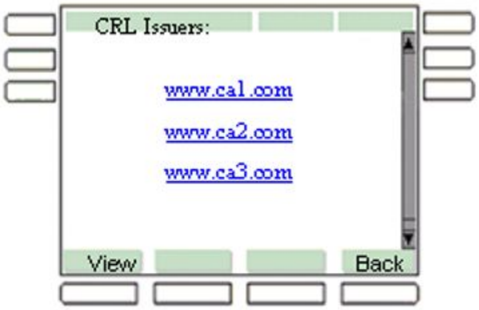
CN: www.deviceserver1.com
SN#: 42:A2:10:24:D0:F9:CE
Usage: EAP-TLS
Expiry: 7/28/2009
Profile Index: 0
Status: OK

Figure 35: Device Certificate details

The administrator can delete the certificate in the "Detailed Mode" by using the Delete Context-sensitive soft key. Deletion does not happen automatically; the IP Deskphone displays a warning confirmation screen.

CRL screen

The CRL screen displays a list of CA issued CRLs stored in the IP Deskphone, as shown in the following figure:



CRL Issuers:

www.ca1.com

www.ca2.com

www.ca3.com

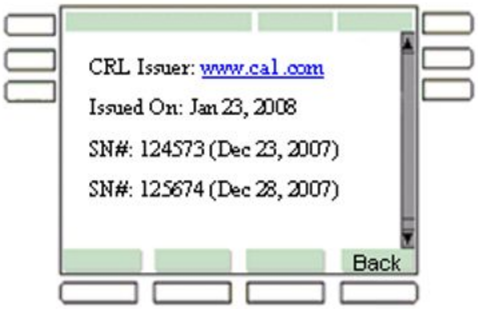
Figure 36: CRL screen

The following table describes the function of the Context-sensitive soft keys for the CRL screen.

Table 77: Context-sensitive soft keys for the CRL screen

Context-sensitive soft key	Action
View	Displays the information on the selected CRL Issuer which includes the following: • CRL Issuer • Issued date • List of serial numbers that belong to the CRL associated with the revocation date.
Back	Returns you to the previous screen.

The following figure is an example of the CRL Details screen for the CRL Issuer www.ca1.com.



CRL Issuer: www.ca1.com

Issued On: Jan 23, 2008

SN#: 124573 (Dec 23, 2007)

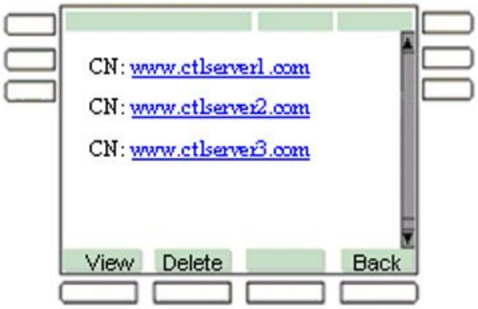
SN#: 125674 (Dec 28, 2007)

Figure 37: CRL details

If you delete a Trusted Anchor Certificate, the CRL issued by the anchor is also deleted.

CTL screen

The CTL screen displays a list of subject Common Name (CN) of the CTL certificates as shown in the following figure:



CN: www.ctlserver1.com

CN: www.ctlserver2.com

CN: www.ctlserver3.com

Figure 38: CTL certificate screen

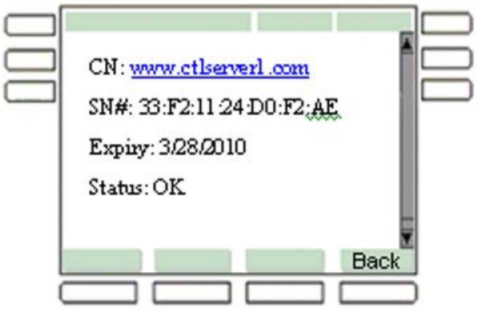
The following table describes the function of the Context-sensitive soft keys for the CTL screen.

Table 78: Context-sensitive soft keys for the CTL screen

Context-sensitive soft key	Action
View	Displays information on the selected certificate which includes the following: • Common Name CN) • Serial Number (SN#) • Expiry Date • Certificate Status (such as, OK or Expired) •
Delete	Displays a warning confirmation. Deletes the CTL.
Back	Returns you to the previous screen.

After you press the View Context-sensitive soft key on the required certificate, information about the certificate you selected appears on the screen.

The following figure is an example of the CTL Certificate Details screen for the certificate www.ctlserver1.com.



CN: www.ctlserverl.com

SN#: 33:F2:11:24:D0:F2:AE

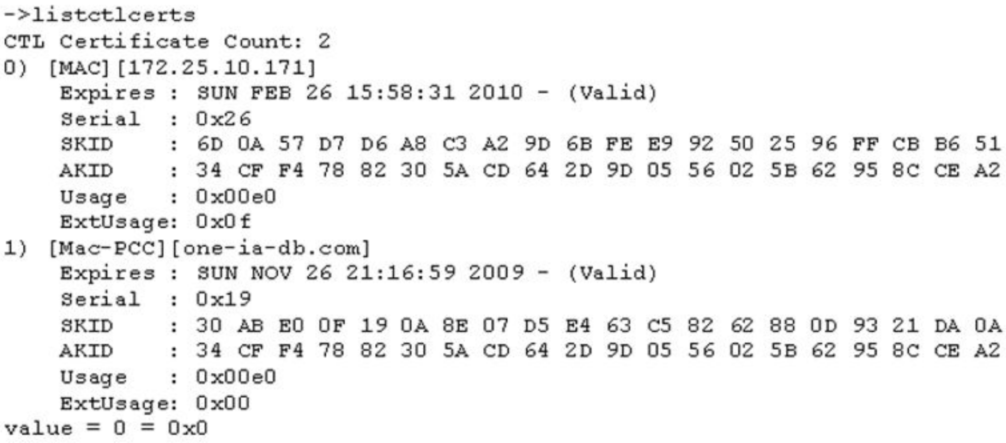
Expiry: 3/28/2010

Status: OK

Figure 39: CTL Certificate details screen

You can use the PDT shell command to view an installed CTL.

The following is an example command with the output of the command.



```
->listctlcerts
CTL Certificate Count: 2
0)  [MAC][172.25.10.171]
     Expires :  SUN FEB 26 15:58:31 2010 - (Valid)
     Serial  :  0x26
     SKID    :  6D 0A 57 D7 D6 A8 C3 A2 9D 6B FE E9 92 50 25 96 FF CB B6 51
     AKID    :  34 CF F4 78 82 30 5A CD 64 2D 9D 05 56 02 5B 62 95 8C CE A2
     Usage   :  0x00e0
     ExtUsage:  0x0f
1)  [Mac-PCC][one-ia-db.com]
     Expires :  SUN NOV 26 21:16:59 2009 - (Valid)
     Serial  :  0x19
     SKID    :  30 AB E0 0F 19 0A 8E 07 D5 E4 63 C5 82 62 88 0D 93 21 DA 0A
     AKID    :  34 CF F4 78 82 30 5A CD 64 2D 9D 05 56 02 5B 62 95 8C CE A2
     Usage   :  0x00e0
     ExtUsage:  0x00
value = 0 = 0x0
```

Figure 40: Example of command output

Important:

The CTL file size must not exceed 20 Kbytes.

EAP Authentication

EAP-enabled networks allow the administrator to ensure that individual devices or users are authorized to access the enterprise's LAN environment.

The following diagram shows the network architecture for 802.1x and EAP.

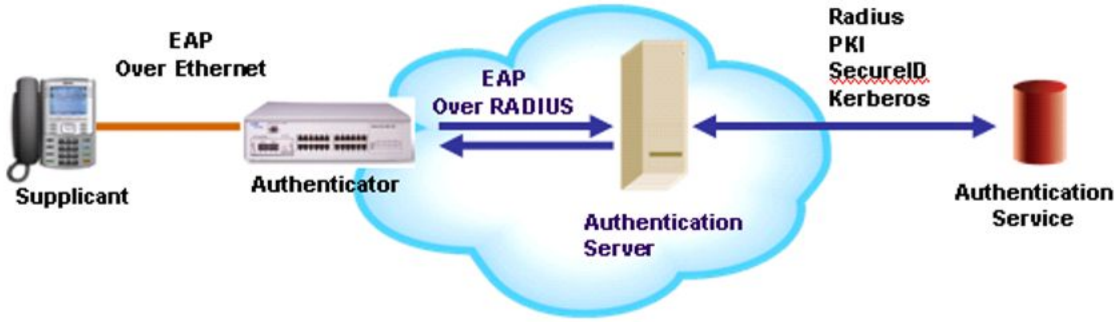


Figure 41: 802.1x and EAP network architecture

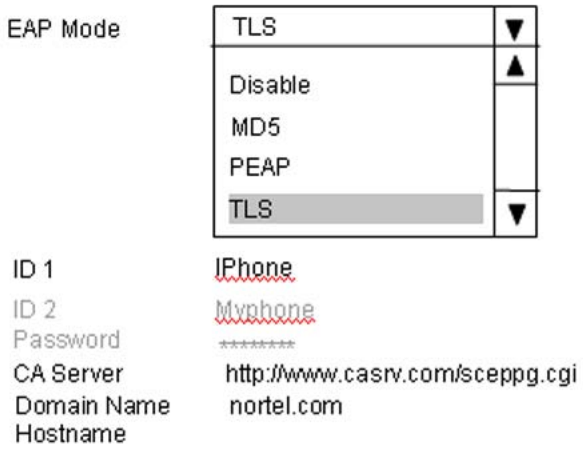
IEEE 802.1x defines three roles:

- a supplicant—an entity that requires access to the network for use of its services.
- an authenticator—the network entry point to which the supplicant physically connects, typically a Layer 2 switch. The authenticator acts as a proxy between the supplicant and the authentication server and controls the access to the network based on the authentication status of the supplicant.
- an authentication server—typically a RADIUS server; performs the actual authentication of the supplicant.

There are three supported EAP methods:

- EAP-MD5
- EAP-TLS
- EAP-PEAP/MD5

The administrator selects the EAP method from the EAP configuration menu, as shown in the following figure:



EAP Mode	TLS
ID 1	IPhone
ID 2	Myphone
Password	********
CA Server	http://www.casrv.com/sceppg.cgi
Domain Name	nortel.com
Hostname	

Figure 42: EAP configuration menu

The administrator can do the following:

- When EAP-MD5 is selected, the administrator is prompted to enter ID1 and Password.
- When EAP-PEAP is selected, the administrator is prompted to enter ID1, ID2 and Password. If the administrator enters only ID1, then ID2 has the same value of ID1.
- When EAP-TLS is selected, the administrator is prompted to enter ID1. If SCEP is used to install the device certificate, the administrator is required to enter the CA Server (URL of the SCEP service), the Domain Name which the IP Deskphone belongs to, and the Hostname (optional).
- When Disable is selected, the existing IDs and passwords are erased.

The following is a list of additional provisioning file parameters for EAP support in addition to the UI parameters on the Device Settings screen

Table 79: EAP Provisioning Parameters

Parameter	Purpose	Default	Allowed
EAP	EAP mode	DISABLED	DISABLED/MD5/ PEAP
EAPID1	Device ID1	Empty	String (4 to 20 characters)
EAPID2	Device ID2	Empty	String (4 to 20 characters)
EAPPWD	Password	Empty	String (4 to 12 characters)

EAP Authentication failures are logged using event 1033.

The following is an example of a TLS authentication failure

```
1033 [Minor][FRI MAY 15 13:48:06 2009][10223][n:/fw/build/../bsp/
vxWorks/common/dot1x/Supplicant/moceap_tls.c:147] - EAP-TLS Failed
to Authenticate
```

The following sections describe the behavior of each method:

EAP Disabled

EAP disabled is the factory default setting. The IP Deskphone does not send a message to the authenticator upon startup, and normal network access is attempted. If the IP Deskphone receives a Request-Identity message from the Layer 2 switch, the Request-Identity is ignored. If the Layer 2 switch requires 802.1x authentication, the IP Deskphone is blocked from the network, and the administrator must enable the EAP feature on the IP Deskphone and configure a DeviceID and Password (if required) to access the network after the IP Deskphone is successfully authenticated. Or, the administrator can plug the IP Deskphone to an EAP disabled port on the Layer 2 switch.

EAP-MDS

EAP-MD5 allows the IP Deskphone to authenticate to the RADIUS server before the IP Deskphone can access the network. This procedure requires a user ID and password. If the IP Deskphone fails to authenticate to the RADIUS server, the IP Deskphone displays a "EAP Authenticate-Fail" message, and the IP Deskphone cannot access the network.

EAP-TLS

EAP-TLS allows the IP Deskphone to authenticate to the RADIUS server before the IP Deskphone can access the network. This procedure requires a user ID, root certificate, and device certificate. The root and device certificates must be installed on the IP Deskphone before using this feature. The customer root certificate can be installed using SCEP or SIP configuration file. For more information, see Root certificate installation on page 199 and Table 71: SCEP provisioning parameters on page 209.

The device certificate can be installed using one of two methods:

- SCEP on page 207
- PKCS 12 download on page 209

If the IP Deskphone fails to authenticate to the RADIUS server or to install the required certificates, the IP Deskphone displays a "EAP Authenticate-Fail" message, and the IP Deskphone cannot access the network.

EAP-PEAP

EAP-PEAP allows the IP Deskphone to authenticate to the RADIUS server before the IP Deskphone can access the network. This procedure requires a user ID1, root certificate, user ID2, and password. EAP-PEAP is the outer authentication protocol that requires a user ID1 and root certificate to establish a TLS channel. EAP-MD5 is the inner authentication protocol that requires a user ID2 and password to pass through this channel in a secure mode. The customer root certificate can be installed using SCEP or SIP configuration file. For more information, see Root certificate installation on page 199.

If the IP Deskphone fails to authenticate to the RADIUS server or to install the required certificates, the IP Deskphone displays a "EAP Authenticate-Fail" message, and the IP Deskphone cannot access the network.

EAP Re-authentication

The re-authentication process proceeds in the background without disturbing the ongoing operation of the IP Deskphone. If the re-authentication fails or times out, the IP Deskphone becomes inoperable. Re-authentication interval is controlled by the Layer 2 switch re-authentication interval parameter. The minimum supported re-authentication interval when EAP-MD5 and EAP-PEAP are configured is 10 seconds; for EAP-TLS, the minimum interval is 20 seconds.

Provisioning configuration file download

Securely download provisioning configuration files through HTTPS.

Provisioning configuration files download through HTTPS

The IP Deskphone can contact a provisioning server and download an 12xxSIP.cfg file to identify additional files and protocols used. When a file is identified, and the protocol specified in the "protocol" parameter is HTTPS, the IP Deskphone contacts the target server and negotiates a TLS connection. Then, the IP Deskphone downloads the specified file and terminates the connection.

HTTP connection over TLS is established by using single or mutual authentication.

Single Authentication

A server certificate, user name, and password are required to establish TLS connection between the IP Deskphone and the provisioning server. The server certificate must be signed by a certificate authority. The IP Deskphone uses the server certificate to validate the identity of the provisioning server that the IP Deskphone is connected to; the provisioning server uses the user name and password to authenticate the IP Deskphone. The IP Deskphone must be preloaded with the root certificate used in signing the server certificate. The root certificate is downloaded to the IP Deskphone by connecting to a provisioning server through EAP-MD5, and using one of the insecure protocols supported by the IP Deskphone, such as HTTP, TFTP or FTP. EAP-MD5 ensures that the connection between the IP Deskphone and the provisioning server is secure. The user name and password are required to authenticate the IP Deskphone to the provisioning server and must be loaded in a secure manner before the IP Deskphone establishes the HTTPS connection with the provisioning server. There is no mechanism for getting a user name and password on the IP Deskphone in a secure "no-touch" manner; the IP Deskphone must be deployed to a secure network where the TFTP download of insecure files is not transmitted over an insecure network.

Mutual Authentication

A device certificate and server certificate are required to establish TLS connection between the IP Deskphone and the provisioning server. The server certificate must be signed by a certificate authority. The IP Deskphone uses the server certificate to validate the identity of the provisioning server that the IP Deskphone is connected to; the provisioning server uses the device certificate to validate the identify of the IP Deskphone. The IP Deskphone must be preloaded with the root certificate used in signing the server certificate. The root certificate is downloaded to the IP Deskphone by connecting to a provisioning server through EAP-MD5, and using one of the insecure protocols supported by the IP Deskphone, such as HTTP, TFTP or FTP. EAP-MD5 ensures that the connection between the IP Deskphone and the provisioning server is secure. The administrator can use the existing device certificates, such as EAP-TLS or SIP-TLS device certificate, instead of having a special device certificate for HTTPS, to establish mutual authentication. For details about device certificate installation and certificate profiles, see Device certificate installation on page 201.

Security and error logs

You can access the Security Log and the Error Log to view errors and failures that may have occurred during the operation of the IP Deskphone.

Before you can access the Security and Error Logs, you must configure the Security Policy file with the following parameter:

SECURITY_LOG_UI_ENABLE YES

If configured as yes, you can access the Security and Error Logs from the Network screen, by choosing Device Settings, Diagnostics, and then Security and Error Logs.

The Security and Error Logs are stored in the Logs folder. To access the Security and Error Logs, select File Manager > Logs folder, and then press the Services key.

The logs main menu lets you choose one of the following options:

1. Security Log
2. Error Log

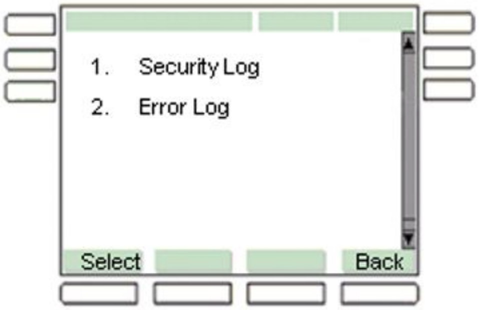
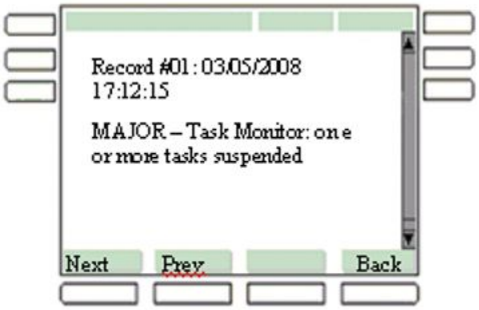


Figure 43: Logs main menu

When the user selects a log file, the screen displays each log item on a full screen, as shown in the following figure:


Record #01 : 03/05/2008
17:12:15

MAJOR – Task Monitor: on e
or more tasks suspended

Figure 44: Log item screen

The following table describes the function of the Context-sensitive soft keys for the log item screen.

Table 80: Context-sensitive soft keys for the log item screen

Context-sensitive soft key	Action
Next	Navigates to the next log entry.
Prev	Navigates to the previous log entry.
Back	Returns you to the Logs main menu.

Security policy file updates

The security policy file contains a set of rules for certificate-based authentication on the IP Deskphone. The rules include the following:

- CERT_ADMIN_UI_ENABLE Determines if the Certificate Administration user interface is enabled on the IP Deskphone. The acceptable values are YES and NO; the default value is NO.
- SECURITY_LOG_UI_ENABLE Determines if the Security Log user interface is enabled on the IP Deskphone. The acceptable values are YES and NO; the default value is NO.
- KEY_SIZE The default size used when generating keys on the IP Deskphone. Acts as the minimum allowed key size that should be enforced when loading certificates from the IP Deskphone. The acceptable values are:
    - KEY_SIZE_1024
    - KEY_SIZE_1536
    - KEY_SIZE_2048

    The default value is KEY_SIZE_1024.
- KEY_ALGORITHM The preferred key generation algorithm. The acceptable value is:

  KEY_ALG_RSA
- DWNLD_CFG_SIGNING defines if configuration files are forced to be signed when a customer certificate is installed.
    - NO - automatically accept the downloaded file without authentication
    - YES - file must be signed and fully authenticated

    The default is NO.
- CUST_CERT_ACCEPT_VAL_NO_CHECK is added to the existing values (VAL_NO_MANUAL, VAL_MANUAL_A, VAL_MANUAL_B.

  The default value is VAL_MANUAL_A).
- SEC_POLICY_ACCEPT is for Security Policy File acceptance (VAL_MANUAL_A, VAL_MANUAL_B.

  The default value is VAL_MANUAL_A)
- SIGN_SIP_CONFIG_FILES Overrides the file signing of a file, such as the device configuration file and the dial plan file. You cannot override the file signing of the Security Policy and Customer Certificates. The acceptable values are:
    - YES—Signing is required.
    - NO—No authentication check is performed.

    The default value is NO.

- FP_PRESENTED If the resource file is not signed and if there are no customer certificates, then you are prompted with a Finger Print display with the option to accept or reject
- FP_ENTERED If the resource file is not signed and if there are no customer certificates, then you must manually enter the Finger Print value and then select Accept.
- SUBJ_ALT_NAME_CHECK_ENABLE Checks the Subject Alternative Attribute in the presented certificate. The acceptable values are YES and NO. The default value is NO.
- CERT_EXPIRE is for certification expiration policy. The acceptable values are:
  - DELETE_CERT
  - LOG_EXPIRE
  - NO_EXPIRE_LOG
- DWNLD_CFG_ACCEPT defines how TFTP configuration authenticates when there are no customer certificates on the phone. The default value is VAL_ACCEPT The acceptable values are:
  - VAL_ACCEPT
  - VAL_MANUAL_A
  - VAL_MANUAL_B
- DWNLD_CFG_SIGNING defines if configuration files are forced to be signed when a customer certificate is installed. The default is NO. The acceptable values are:
  - NO - automatically accept the downloaded file without authentication
  - YES - file must be signed and fully authenticated

Changes made to the security policy file have an entry in the security log file.

SECURITY_POLICY_PARAM_CHANGE	0x1055

The security log file stores only the non-sensitive information. For example, if the password is changed, the security log file indicates this change without storing the password value.

You can use the PDT shell command to view the output of the security policy command.

The following is the output of the securitypolicy command from the PDT shell.

```
-> securitypolicy

CUST_CERT_ACCEPT = VAL_MANUAL_A

SEC_POLICY_ACCEPT = VAL_MANUAL_A

SIGN_SIP_CONFIG_FILES = NO

CERT_EXPIRE = DELETE_CERT

SEC_POLICY_TEXT = YES

AUTO_PRV_ACCEPT = VAL_ACCEPT
```

```
DWNLD_CFG_ACCEPT = VAL_ACCEPT
AUTO_PRV_SIGNING = NO
DWNLD_CFG_SIGNING = NO
CERT_ADMIN_UI_ENABLE = YES
SECURITY_LOG_UI_ENABLE = YES
KEY_SIZE = KEY_SIZE_1024
KEY_ALGORITHM = KEY_ALG_RSA
TLS_CIPHER = RSA_WITH_AES_256_CBC_SHA
SUBJ_ALT_NAME_CHECK_ENABLE = NO
FTP_PASSWORD = ****
```

Certificate Admin option in the user interface

The following procedure provides the steps to view the Certificate Admin option in the user interface.

Viewing the Certificate Admin option in the user interface

1. Create a text file, for example SecurityPolicy.txt.
2. Add CERT_ADMIN_UI_ENABLE YES in the text file.
3. Sign the file using a signing certificate. For example, SecurityPolicy.txt.sig file is created.
4. Download the file using [SEC_POLICY] section in the 11xxeSIP.cfg file. An example of the SEC_POLICY section is as follows:

   ```
   [SEC_POLICY]
   DOWNLOAD_MODE_FORCED
   PROTCOL HTTP
   ```

Installation

This section describes the following:

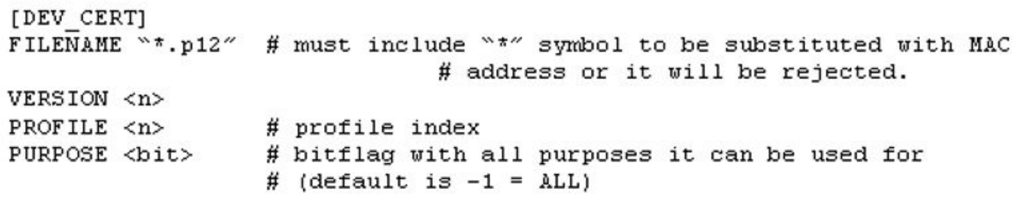
- Device certificate installation using PKCS#12
- CTL download

Device Certificate Installation

SCEP and PKCS#12 are two methods used to install device certificates. If SCEP is used to install device certificates, see SCEP on page 207 for more information.

The following describes the process of using PKCS#12 to install device certificates.

1. The administrator adds a [DEV_CERT] section to 12xxSIP.cfg to let the IP Deskphone import a PKCS#12 file. The following is an example of the format of the [DEV_CERT] section:



```
[DEV_CERT]
FILENAME “*.p12”  # must include “*” symbol to be substituted with MAC
                              # address or it will be rejected.
VERSION <n>
PROFILE <n>       # profile index
PURPOSE <bit>     # bitflag with all purposes it can be used for
                  # (default is -1 = ALL)
```

Figure 45: Example of [DEV_CERT] section

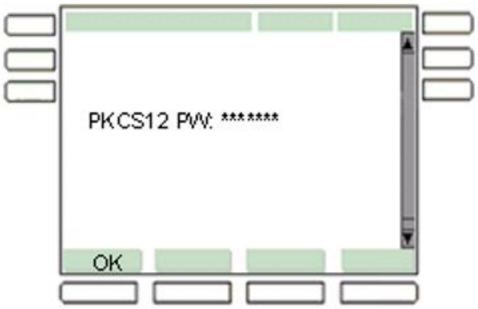
- The "*" pointed by the FILENAME attribute is substituted with the IP Deskphone MAC address before the IP Deskphone requests the PKCS#12 file. If the IP Deskphone has multiple PKCS#12 files, the administrator must add another ID beside "*". This ID can be the profile index.
- The VERSION attribute determines if the file should be downloaded by comparing this VERSION with the VERSION stored in the corresponding device certificate profile.
- The PROFILE attribute points to the device certificate profile index. The certificate profile index identifies the file name where the profile is stored in the IP Deskphone memory (SFS), and identifies the device certificate profile.
- The PURPOSE attribute identifies device certificate usage. The purpose attribute is a bit mask that lets a device certificate be used for multiple purposes; for example, sharing of device certificates. These purposes can be:

  - EAP-TLS

  - SIP-TLS

  - HTTPS (optional)

2. After 12xxSIP.cfg is downloaded to the IP Deskphone from the provisioning server, the IP Deskphone executes the [DEV_CERT] section and downloads the PKCS#12 file.
3. After the PKCS#12 file is downloaded, the IP Deskphone prompts the administrator to enter the PKCS#12 protected password as shown in the following figure.



PKCS12 PW: *******

Figure 46: PKCS12 password prompt

> **Important:**
>
> The password can be empty, but the use of an empty password is not recommended except under very controlled conditions.

4. After the password is validated, the IP Deskphone extracts the private key and device certificate from the PKCS#12 file.
5. The IP Deskphone validates the device certificate to ensure that the device certificate is signed by a trusted CA, is not revoked, and that the key size meets the minimum requirement.
6. If the device certificate is validated correctly, the IP Deskphone stores the device certificate and the private key in the IP Deskphone memory (SFS) in the device certificate profile specified in the [DEV_CERT] section.

CTL download

This section describes the process of downloading a CTL file on the IP Deskphone.

1. The administrator adds the [CTL] section to 12xxSIP.cfg to allow the IP Deskphone to download a CTL file. The following is an example of the format for the [CTL] section:

   ```
   [CTL] DOWNLOAD_MODE AUTO PROTOCOL HTTPS FILENAME ctl.pem
   ```

2. After 12xxSIP.cfg is downloaded to the IP Deskphone from the provisioning server the IP Deskphone executes the [CTL] section and downloads the CTL file.
3. After the CTL file is downloaded, the IP Deskphone validates the CTL file to ensure that the CTL file is signed by a trusted entity. If the CTL file is validated correctly, the CTL file is stored in the IP Deskphone memory (SFS).

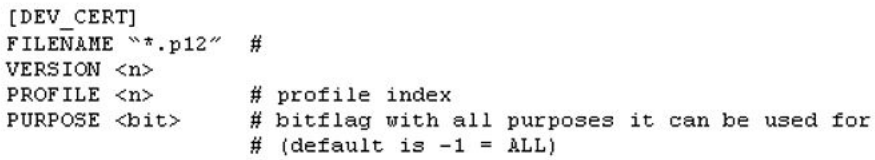
Upgrade and rollback tasks

The IP Deskphone loaded with secure software cannot downgrade to a previous insecure framework.

SIP configuration file (12xxSIP.cfg)

PKCS12 Import

The [DEV_CERT] section is added to 12xxSIP.cfg to let the IP Deskphone import a PKCS#12 file. The following is an example of the format of the [DEV_CERT] section:



```
[DEV_CERT]
FILENAME “*.p12”  #
VERSION <n>
PROFILE <n>       # profile index
PURPOSE <bit>     # bitflag with all purposes it can be used for
                  # (default is -1 = ALL)
```

Figure 47: Example of [DEV_CERT] section

- The "*" pointed by the FILENAME attribute is substituted with the IP Deskphone MAC address before the IP Deskphone requests the PKCS#12 file. If the IP Deskphone has multiple PKCS#12 files, the administrator must add another ID beside "*". This ID can be the profile index.
- The VERSION attribute determines if the file should be downloaded by comparing this VERSION with the VERSION stored in the corresponding device certificate profile.
- The PROFILE attribute points to the device certificate profile index. The certificate profile index identifies the file name where the profile is stored in the IP Deskphone memory (SFS), and identifies the device certificate profile.
- The PURPOSE attribute identifies device certificate usage. The purpose attribute is a bit mask that lets a device certificate be used for multiple purposes; for example, sharing of device certificates. These purposes can be:
    - EAP-TLS
    - SIP-TLS
    - HTTPS (optional)

CTL download

The [CTL] section is added to 12xxSIP.cfg to allow the IP Deskphone to download a CTL file from a provisioning server. The following is an example of the format for the [CTL] section:

```
[CTL] DOWNLOAD_MODE AUTO PROTOCOL HTTPS FILENAME ctl.pem
```

Customer root certificate download

The [USER_KEYS] section is added to 12xxSIP.cfg to allow the IP Deskphone to download a customer root certificate from a provisioning server. The following is an example of the format for the [USER_KEYS] section:

```
[USER_KEYS] DOWNLOAD_MODE AUTO PROTOCOL HTTPS FILENAME custroot.pem
```

Security policy file

The security policy file defines a set of rules to determine the required actions taken by the IP Deskphone. The following is an example of security policy file rules and default actions:

```
CERT_ADMIN_UI_ENABLE NO SECURITY_LOG_UI_ENABLE NO KEY_SIZE 1024
KEY_ALGORITHM KEY_ALG_RSA TLS_CIPHER RSA_WITH _AES_256_CBC_SHA
```

The format of the security policy file, as shown in the preceding example, is parameter-value paired. The parameter name and value are separated by a space.

Diagnostic logs

All EAP failures are logged in the security log which include the following EAP error messages:

EAP_MD5_AUTH_FAILURE	0x1030
EAP_INVALID_DEVICE_CERTIFICATE	0x1031
EAP_INVALID_ROOT_CERTIFICATE	0x1032
EAP_TLS_AUTH_FAILURE	0x1033
EAP_PEAP_AUTH_FAILURE	0x1034

The following is a list of certificate-related events and failures logged in the Security Log.

SLC_AVAYA_CERTIFICATE_IMPORTED	0x0006
SLC_SERVICE_PROVIDER_CERTIFICATE_IMPORTED	0x0007
SLC_AVAYA_CERTIFICATE_REVOKED	0x0008
SLC_SERVICE_PROVIDER_CERTIFICATE_REVOKED	0x0009
SLC_AVAYA_CERTIFICATE_EXPIRED	0x000A
SLC_SERVICE_PROVIDER_CERTIFICATE_EXPIRED	0x000B
SLC_CERTIFICATE_DELETED	0x000C
SLC_CRL_IMPORTED	0X000D
SLC_OLDER_CRL_REMOVED	0x000E
SLC_FACTORY_DEFAULTS_RESTORED	0x000F
SLC_DEVICE_CERTIFICATE_CREATED	0x0010
SLC_CRL_SIGNATURE_REJECTED	0x0011
SLC_CTL_CERTIFICATE_EXPIRED	0x0012
SLC_AVAYA_CERTIFICATE_DELETED	0x0013
SLC_SERVICE_PROVIDER_DELETED	0x0014
SLC_CTL_DELETED	0x0015
SLC_CRL_DELETED	0x0016
SLC_DEVICE_CERTIFICATE_DELETED	0x0017
SLC_DEVICE_CERTIFICATE_REVOKED	0x0018
SLC_DEVICE_CERTIFICATE_EXPIRED	0x0019
SLC_CTL_EXPIRED	0x0020
SLC_CTL_DOWNLOAD_ERROR	0x0021

The following is a list of minor errors that are logged in the Security Log.

SLC_AVAYA_CERTIFICATE_EXPIRED_AUTH	0x1002
SLC_SERVICE_PROVIDER_CERTIFICATE_EXPIRED_AUTH	0x1003

SLC_PROVIDER_CERTIFICATE_IN_AVAYA_KEYS_FILE	0x1004
SLC_PKI_MGMT_INIT_FAILURE	0x1005

The following is the Security Policy parameter change event.

SECURITY_POLICY_PARAM_CHANGE	0x1055

Any changes made to the security policy file has an entry in the security log file. For more information, see Security policy file updates on page 228.

Fault management behavior

Authentication failures are indicated by a failure message on the IP Deskphone screen and are reported to the error log files. The administrator can view the security logger by using the PDT or the security log viewer. For more information, see Security and error logs on page 226.

The following is a list of authentication failure messages that appear on the IP Deskphone screen when a failure occurs during the operation of the IP Deskphone:

- EAP Authenticate-Fail—happens when the IP Deskphone fails to authenticate to an authentication server; the message applies for the three EAP methods: EAP-MD5, EAP-PEAP, and EAP-TLS.
- EAP Authenticate-Timeout—happens after the third time the IP Deskphone fails to authenticate to an authentication server and the IP Deskphone is connected to an EAP disabled port on the Layer 2 switch.

For EAP failures logged in the security log, see Diagnostic logs on page 234.

Chapter 24: Security

This section specifies the behavior of the following security features:

- SIP over TLS
- Connection persistence
- SRTP
- SFTP
- SSH

SIP over TLS

To avoid security problems such as message integrity attacks, SIP over TLS uses Transport Layer Security (TLS) to provide secure communication between the Avaya 1100 Series IP Deskphone and the SIP proxy.

Transport Layer Security (TLS) protects SIP signaling traffic. It sits on top of the Transmission Control Protocol (TCP), the preferred default protocol for SIP traffic. You can use TLS with a user name and password to provide a means of server-only authentication. IP Deskphone-specific Public Key certificates can provide even stronger mutual-authentication of both the server and the IP Deskphone.

Using SIP over TLS protects SIP messages on a hop-by-hop basis. To achieve complete end-to-end security through the use of TLS, each element involved in the system must also be capable of securing SIP traffic using TLS.

Connection persistence

Connection persistence allows the IP Deskphone to establish a connection and monitor the connection for failure by using "keep-alive requests.

The IP Deskphone establishes connection with the proxy using the commonly accepted ports. Periodically, based on a configured timer value, the IP Deskphone issues a request to the server to verify that the connection with the server at the TCP level is still active. When the IP Deskphone discovers that the keep-alive packet has not been answered, it attempts to reestablish a connection with the proxy. If this is successful, the IP Deskphone reregisters with the proxy (and sends a new subscription requests where appropriate). If it is not possible to reestablish the connection, the IP Deskphone falls back into a state where connection attempts are tried periodically based on random, but increasing time periods, in order to give the server adequate time to recover.

SSH and secure file transfer

The Secure Shell Handler (SSH) is a widely-used protocol for providing secure logon access to run commands remotely. To establish a connection, you must access the SSH-capable client, and know the user name and password that is configured on the IP Deskphone through the use of the provisioning system.

Secure File Transfer Protocol (SFTP) lets the administrator securely log on to the IP Deskphone (using the common user name and password shared with SSH/PDT). After you logon, the IP Deskphone displays a list of files on the flash file that you can transfer.

SSH and SFTP

The following table provides a list of SSH and SFTP configuration parameters.

Parameter	Description	Default value	Boundaries
Enable SSH	Enables the SSH server on the IP Deskphone for secure shell access.	Not checked (off)	Not checked (off) Checked (on)
Enable SFTP	Enables the SFTP server on the IP Deskphone for secure FTP access. SSH must be enabled for SFTP to be enabled	Not checked (off) (appears dimmed until SSH is enabled)	Not checked (off) Checked (on)
User ID	The User ID that must be entered when connecting to the IP Deskphone SSH or SFTP.	None	Non-null string Maximum: 49 characters

Parameter	Description	Default value	Boundaries
Password	The password that must be entered when connecting to the IP Deskphone through SSH, SFTP.	None	Non-null string Maximum: 49 characters

UI Properties for Device Settings SSH and SFTP parameters are as follows:

- The User ID field is empty and the Password field displays "****" when both SSH and SFTP are disabled and applied.
- The user can enable SSH or SFTP.
- The user must provide a valid user ID and password when the User ID field is empty, and an application (SSH or SFTP) is selected. If a valid user ID and valid password are not provided, and the user presses the **Apply** context-sensitive soft key, one of the following error message appears:
    - `Error: User ID size: 4-12` – appears if a valid user ID is not provided.
    - `Error: Password size: 4-12` – appears if a valid password is not provided.
    - `Error: User ID size: 4-12 Error: Password size: 4-12` – appear if both a valid user ID and a valid password are not provided.

TCP/TLS operation overview

TCP is the alternative protocol the IP Deskphone uses when sending and receiving SIP requests. Avaya recommends TCP for Avaya SIP-enabled entities.

When a server initiates a TCP or TLS connection to the IP Deskphone, the connection only lasts as long as the server chooses to keep the connection open; a persistent connection is not maintained by the IP Deskphone.

How the IP Deskphone uses TCP

TCP is a connection-based protocol, which means the IP Deskphone must first establish a connection with a target. This is done using a three-way handshake. After the handshake process is complete and a connection is made, the IP Deskphone can send data over the TCP connection. The data, which makes up a SIP request, can now be sent and received by either side of the communication.

How the IP Deskphone uses TLS

Transport Level Security (TLS) is a protocol for establishing a secure connection between two end-points. After a connection is established using TCP, TLS negotiates the cryptographic parameters used to secure the traffic that is sent over that connection. TLS, Public Key Cryptography, and X.509 certificates provide either mutual or server authentication.

- Mutual authentication occurs when both the client and the server have public key certificates assigned, that are used during the TLS handshake, to validate the identity of both communicating parties. Both the server and the end point device certificates are "signed" by well-known trusted certificate authorities.
- Server authentication occurs when a server has a certificate signed by a certificate authority. The certificate is only used for the client to validate the identity of the server it is connected to. After the TLS connection is established, the server can identify the IP Deskphone through a user name and password.

How TLS impacts SIP

TLS impacts SIP in the following ways:

- URIs – contain transport parameters used to indicate the preferred method of contact. For example,

  ```
  Contact: Bob<sip:bob@company.com;transport=tls>
  ```

  Important:

  A transport parameter of TLS indicates that the server or client prefers TLS to be used for communication.

  SIP Software Release 3.2 adds transport=tls to the contact header when using TCP or TLS.
- VIA header – contains the transport protocol used to send a request. For example, Via: SIP3.2/TLS`bob.company.com;....;alias`

  The IP Deskphone attempts to downgrade the allowed protocols if connection attempts are made and fail. In order to avoid the IP Deskphone using an unsecure protocol, only TLS is enabled.

  The order of preference for protocols is always: TLS, TCP, and UDP.

  You must enable the SIP TLS Listening port for incoming TLS connections to be made.

Certificate requirements

For the IP Deskphone to validate that the server certificate provided by the TLS-enabled proxy matches the connected address, the certificate must contain the IP Addresses of the IP Deskphone.

The server certificate has a Subject Alternative Name field, which contains the IPv4 and IPv6 IP addresses that correspond with the proxy. For example:

```
subjectAltName=IP:192.168.100.100subjectAltName=IP:
2001:0db8:0000:0000:0000:0000:1428:5 7ab
```

Important:

The IP Deskphone must have a device certificate loaded. If the device certificate is not loaded, the IP Deskphone fails to establish a TLS connection with the system.

IP Deskphone configuration

The following table lists the various security parameters for the IP Deskphone.

Table 81: Provisioning parameters summary

Parameter	Purpose	Default	Allowed
SERVER_TCP_PORT1_1 SERVER_TCP_PORT1_2 SERVER_TCP_PORT2_1 SERVER_TCP_PORT2_2 SERVER_TCP_PORT3_1 SERVER_TCP_PORT3_2 SERVER_TCP_PORT4_1 SERVER_TCP_PORT4_2 SERVER_TCP_PORT5_1 SERVER_TCP_PORT5_2	Configures the TCP and TLS ports used when connecting to the SIP domain.	TCP: 5060 TLS: 5061	Integer

Parameter	Purpose	Default	Allowed
SERVER_TLS_PORT1_1 SERVER_TLS_PORT1_2 SERVER_TLS_PORT2_1 SERVER_TLS_PORT2_2 SERVER_TLS_PORT3_1 SERVER_TLS_PORT3_2 SERVER_TLS_PORT4_1 SERVER_TLS_PORT4_2 SERVER_TLS_PORT5_1 SERVER_TLS_PORT5_2			
SIP_UDP_PORT SIP_TCP_PORT SIP_TLS_PORT	Configures the local SIP listening ports. After you change the listening ports parameters through the Check For Updates functionality, you must restart the IP Deskphone to apply the modified values.	UDP: 5060 TCP: 5060 TLS: 5061	Integer
CONN_KEEP_ALIVE REGISTER_RETRY_TIME REGISTER_RETRY_MAX TIME	Configuration values that affect connection persistent.	<add link to table.	See Managing connection persistence on page 250.
SRTP_ENABLED SRTP_MODE	SRTP configuration values.	NoBE-2MLines	BE-2MLines BE-CapNeg SecureOnly
SRTP_CIPHER_1 SRTP_CIPHER_2	Allows configuration of the preferred order for SRTP cipher offers.	AES_CM_128_HMAC_SHA1_80, AES_CM_128_HMAC_SHA1_32	AES_CM_128_HMAC_SHA1_32 AES_CM_128_HMAC_SHA1_80 None
LOGIN_NOTIFY	Configures whether or not the login banner appears after a successful logon.	Off	OffSuccessFailureBoth

Parameter	Purpose	Default	Allowed
LOGIN_NOTIFY_TIME	Configures whether or not the time at which the login success or failure occurred appears.	Not checked	Not checked (off)Checked (on)
SSH	Configuration of the SSH server on the IP Deskphone. The parameter must remain consistent with the current UNIStim design.	NO	YES, NO
SFTP	Configuration of the SFTP server on the IP Deskphone. The parameter must be added, but can remain consistent with SSH.	NO	YES, NO
SFTP_READ_PATTERNS	File extensions allowed to read (get) from the SIP client.	.cfg,.dat	"," separated values. See Note 1. After a change is detected in this parameter, the system resets.
SFTP_WRITE_PATTERNS	File extensions allowed to write (put) from SIP client.	.cfg,.dat	"," separated values. See Note 1 and Note 2. After a change is detected in this parameter, the system resets.
SSHID	Configuration of the SSH and SFTP user ID.	None	See Note 3.
SSHPWD	Configuration of the SSH and SFTP password.	None	See Note 3.
HASHED_ADMIN_PASSW ORD	Indicate whether the Admin Password is hashed or not.	NO	YES, NO
ENALBE_LOCAL_ADMIN_ UI	Configure the availability of the local administration UI on the IP Deskphone.	YES	YES, NO

Parameter	Purpose	Default	Allowed
HASH_ALGORITHM	Hash algorithm.	SHA1	SHA1, MD5
MKI_ENABLE	Use Master Key Identifier (MKI) or not.	NO	YES, NO
ADMIN_UI_ENABLE	Configure the availability of the local administration User Interface on the IP Deskphone.	YES	YES, NO
SECURE_UI_ENABLE	Configure the availability of other sensitive data that you want to hide from the normal end user, such as the IP address, the MAC address on the IP Deskphone information screen, and the FE IP Address and Port on the audio quality details screen.	NO	YES, NO
ADMIN_PASSWORD_EXPI RY	The date that the configured ADMIN_PWD is no longer valid, and a new password must be downloaded from the provisioning server.	Empty	Timestamp
CONTACT_HDR_PORT_CS1K	Configure SIP IP Phones to register to a Communication Server 1000 server using TLS and to be able to receive calls using TLS.		Y, N

Note:

The SFTP file read and write pattern entries must be strictly followed.

The following are examples of valid and invalid formats of SRTP read and write patterns.Example of valid formats:SFTP_READ_PATTERNS: cfg,.rel,.re2,.re3,.dat SFTP_WRITE_PATTERNS: cfg,.txt,.wr1,.wr2

Example of an invalid format:.cfg, .txt

For the SFTP file read and write pattern entries to be valid, there must be no space between the extensions.

Note:

SFTP writes can only be made to the sftpWr folder. You are only allowed to write a file that is 10%, or less, of the available space on the folder.If a file size greater than 10% is written, a write failure occurs, and the system logs the following event:`1042[Minor][TUE JAN 02 19:08:18 2007][353][i:/fw/build/../util/sshapp/sftpS erver.c:691] - File (./sftpWr/lf.wrl) too large to write.`

Note:

If logon failures occur for SSH and SFTP applications, the system logs the following event: `1040[Minor][TUE JAN 02 20:12:14 2007][4189][i:/fw/build/../sshapp/sshServer .c:616] - SSH Authentication Failed.`

Manually configure the IP Deskphone for UDP and TCP

After you enable the administration user interface, you can manually change network settings on the IP Deskphone. You can manually configure the IP Deskphone through the Server Settings menu.

Note:

To meet security requirements, the local administration user interface of the IP Deskphone can be disabled for deployed IP Deskphones. If this is the case then you must manually configure the parameters during initial IP Deskphone configuration or through the provisioning server.

Note:

Disabling the local administration user interface drastically reduces the ability to view or edit the configuration of the IP Deskphone, and almost completely removes the ability to diagnose any communication or configuration errors in the field. However, disabling the local administration user interface increases the security of the IP Deskphone because the user is not able to view the configurations or make changes.

Configuring the domain protocol

1. Press the **Globe** key twice.
2. Using the Navigation key cluster, select **Server Settings**..

3. Select a domain.
4. Enter the admin password (if the UI and password are enabled).
5. Use the Navigation key cluster to scroll through the Domain List screen and select the required configured SIP domain.
6. Press the **Edit** context-sensitive soft key.

Table 82: Listening port parameters

Parameter name	Description	Default value	Boundaries
SIP UDP Port	The listening port on the IP Deskphone for incoming UDP requests.	5060	Min: 1024 Max: 65535 Disabled: 0 (must be non-zero for a TLS-only option)
SIP TCP Port	The listening port on the IP Deskphone for incoming TCP requests.	5060	Min: 1024 Max: 65535 Disabled: 0 (must be non-zero for a TLS-only option)
SIP TLS Port	The listening port on the IP Deskphone for incoming TLS requests.	0	Min: 1024 Max: 65535 Disabled: 0 (must be non-zero for a TLS-only option)

Note:

The configuration of the IP Deskphone for various protocols must be completed for outgoing and incoming connections. For a complete TLS-only option, the outgoing server UDP and TCP protocols must be configured as a non-zero value, and the incoming UDP and TCP listening ports must be configured as a non-zero value.

Using the TLS to connect to the SIP proxy

The IP Deskphone can establish a connection with the proxy after the appropriate configurations are made for the TLS. After the IP Deskphone registers with the SIP Proxy, the user can detect if a secure connection is established by the presence of a security icon (padlock) on the idle screen.

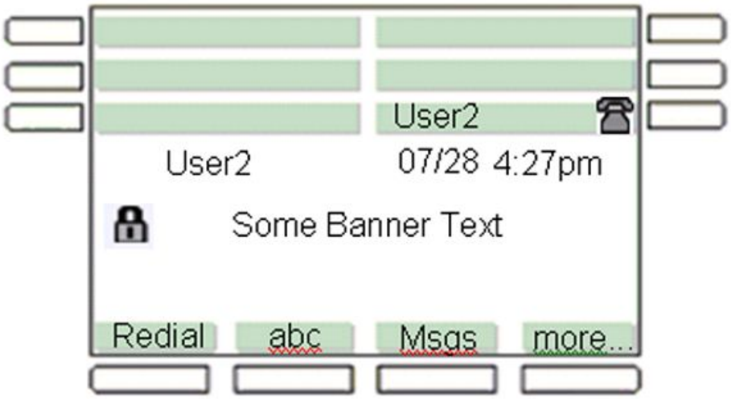


Figure 48: Security icon enabled

Note:

Connecting to the server requires that the IP Deskphone uses, at a minimum, TLS_RSA_WITH_AES_128_CBC_SHA, and as an objective, TLS_RSA_WITH_AES_256_CBC_SHA. Because this is a server-specific configuration, the IP Deskphone must be prepared to handle both. There is no difference in screen indication, regardless of the type of cipher used.

The following table describes the configurations that affect the presence of the security icon on the idle screen of the IP Deskphone.

Configuration	Result	Idle Screen Security Icon Display
Default: UDP + TCP	SIP is unsecured.	No
UDP only	SIP is unsecured.	No
TCP only	SIP is unsecured.	No
TLS only	Connection is only established if SIP is secure.	Yes
UDP + TLS: unsupported	Unsupported.	Unsupported
TCP + TLS	Connection is established with either TCP or TLS.	Yes – only if TLS connection is used No – if fall back to TCP occurs
UDP + TCP + TLS	Connection is established using TCP or TLS, potentially falling back to using only UDP.	Yes – only if TLS connection is established No – if fall back to TCP or UDP occurs
None : unsupported	Unsupported	Unsupported

Unsupported configurations cannot be saved. If the configurations are unsupported, the IP Deskphone displays an error message.

The following is an example of an error message for unsupported configurations:`Unsupported: UDP + TLS`Unsupported: No protocols enabled.

Registration behavior based on configuration settings

The following table describes the behavior of the IP Deskphone when the IP Deskphone is configured to communicate with a server using specific protocols.

Table 83: Registration results based on configuration

Configuration	Description	Expected result	Possible results
IP Deskphone: UDP + TCP Server: UDP + TCP + TLS	The IP Deskphone allows protocols enabled for communication with the server.	The IP Deskphone establishes a connection to the server using TCP.	If the server does not accept incoming requests on TCP, it takes approximately thirty seconds for the initial connection attempt to fail, and then the IP Deskphone attempts to contact the server using UDP. If this connection also fails, the IP Deskphone waits a configured period of time before attempting to reconnect.
IP Deskphone: UDP Server: UDP + TCP + TLS	The IP Deskphone only has UDP enabled for sending requests to the server.	The IP Deskphone registers using UDP as the protocol.	If the IP Deskphone is unable to contact the server, it waits a configured period of time before attempting to reconnect.
IP Deskphone: TCP only Server: UDP + TCP + TLS	The IP Deskphone only has TCP enabled for sending requests to the server.	The IP Deskphone registers using TCP as the protocol.	If the IP Deskphone is unable to contact the server, it waits a configured period of time before attempting to reconnect.
IP Deskphon	The IP Deskphone only has TLS configured for	The IP Deskphone registers using SIP over	If the IP Deskphone is unable to contact the

Configuration	Description	Expected result	Possible results
e: TLS only Server: UDP + TCP + TLS	sending requests to the server. The IP Deskphone must have a device certificate installed if the server is configured for mutual authentication.	TLS. If a device certificate is provisioned, and the server is configured for mutual authentication, then the IP Deskphone provides a certificate during the TLS handshake. Otherwise, server-only authentication is used.	server, it waits a configured period of time before attempting to reconnect.
UDP + TLS: unsupported	Unsupported	Unsupported	Unsupported
IP Deskphone: TCP + TLS Server: UDP + TCP + TLS	The IP Deskphone attempts to contact the server using TLS first, because TLS has higher priority than TCP.	The IP Deskphone registers the same as if it was configured for TLS only.	If the IP Deskphone is unable to connect to the server using TLS, it attempts to connect using TCP. If attempts to connect using TLS and TCP fail, the IP Deskphone waits a configured period of time before attempting to reconnect.
IP Deskphone: UDP + TCP + TLS Server: UDP + TCP + TLS	The IP Deskphone attempts to contact the server using TLS first, because TLS has higher priority than TCP and UDP.	The IP Deskphone registers the same as if it was configured for TLS only.	If the IP Deskphone is unable to connect to the server using TLS, it attempts to connect using TCP. If attempts to connect using TLS and TCP fail, the IP Deskphone attempts to connect using UDP. If attempts using TLS, TCP, and UDP fail, the IP Deskphone waits a configured period of time before attempting to reconnect.
None: unsupported	Unsupported	Unsupported	Unsupported

Note:

The server must be configured with the appropriate protocols enabled for the success condition to be realized. Failure results are possible if the server configuration is changed to disallow protocols.

Managing connection persistence

The IP Deskphone attempts to establish and maintain a persistent connection with the proxy when TCP and TLS are active protocols. After this connection is established, the IP Deskphone sends all outgoing connections over this persistent connection.

SIP IP Deskphones and servers, which use UDP to communicate, listen for incoming connections on known ports, and originate each request on a randomly selected UDP port. Even if TCP is used, new requests can potentially be sent using a new source port unless the connection between the IP Deskphone and proxy is kept active.

Connection persistence does the following:

- Keeps a connection established between a client and the outgoing proxy.
- Reuses the open connection for future incoming and outgoing requests.

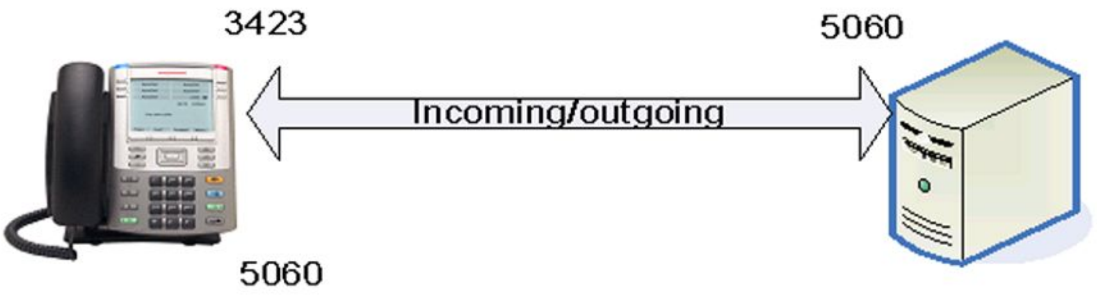


Figure 49: Incoming/Outgoing with connection reuse

When using UDP, an IP Deskphone behind a firewall must periodically send a request to the server to maintain an open pinhole in the firewall so that the server can contact the IP Deskphone when sending requests.

When using TCP/TLS and connection persistence, it is not necessary to send a SIP_PING to the server in order to keep a pinhole alive, and the keep-alive mechanism is reduced to a method which involves significantly less overhead.

The following figure demonstrates how critical it is that the server can communicate directly with the IP Deskphone through the use of the established TCP connection because it has no way of getting through the firewall in order to contact port 5060 on the IP Deskphone.

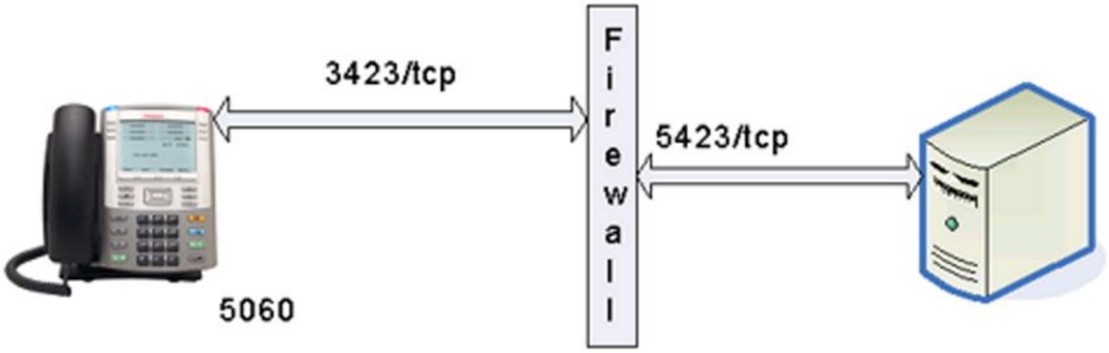


Figure 50: Connection reuse and a firewall

Table 84: Connection timers definitions and allowed values

Parameter name	Description	Default value	Boundaries
OS Keep-alive only	Selecting this value causes the OS TCP Keep-alive functions to be used instead of the CRLF ping/pong mechanism. Some system deployments may prefer the lighter weight TCP keep-alive	Not checked	Checked
Keep-alive	This is a value, measured in seconds, that the IP Deskphone uses when a connection to the server is established using TCP or TLS. The IP Deskphone periodically sends a packet to the server, which contains a pair of CRLF, to ensure the server is responding.	30	Min: 15 Max: 1800
Register Retry	When a connection failure occurs, this value in seconds is how long the IP Deskphone waits before attempting to reregister with the proxy.	30	Min: 30 Max: 1800
Register Max Retry	After a failure to reconnect with the proxy, the IP Deskphone increases the amount of time that it	1800	Min: 600 Max: 1800

Parameter name	Description	Default value	Boundaries
	waits for the next registration retry attempt. This value, measured in seconds, is the maximum value that the IP Deskphone waits in between retry attempts		

SRTP

Secure Real-time Transport Protocol (SRTP) encrypts the Real-time Transport Protocol (RTP) traffic between two end-points to achieve full security for the media path.

Security Descriptions for the Session Description Protocol (SDESC) (RFC4586) defines a mechanism to transmit the necessary cryptographic parameters between two end-points. SRTP is initiated when Secure Real-time Transport Control Protocol (SRTCP) allows both sides of a conversation to agree on the keys you can use to encrypt or decrypt the messages that are transmitted.

Media security — SRTP

Secure RTP (SRTP) encrypts the media path between two end-points. After both end-points agree on the necessary parameters to encrypt and decrypt audio packets, the voice path between them is established.

SRTP is configured on the IP Deskphone to provide multiple levels of protection.

The following table highlights the two cipher suites that are used and their related parameters.

Table 85: SRTP properties

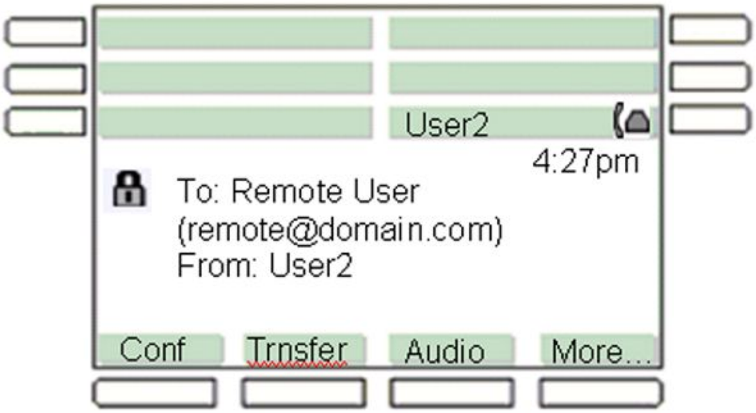
Parameter	AES_CM_128_HMAC_SHA1_80	AES_CM_128_HMAC_SHA1_32
Master key length	128 bits	128 bits
Master salt length	112 bits	112 bits
SRTP lifetime	2^48 packets	2^48 packets

Parameter	AES_CM_128_HMAC_SHA1_80	AES_CM_128_HMAC_SHA1_32
SRTCP lifetime	2^31 packets	2^31 packets
Cipher	AES Counter Mode	AES Counter Mode
Encryption key	128 bits	128 bits
MAC	HMAC-SHA1	HMAC-SHA1
SRTP auth. tag	80 bits	32 bits
SRTCP auth. tag	80 bits	80 bits
SRTP auth. key len.	160 bits	160 bits
SRTCP auth. key len.	160 bits	160 bits

Call security is identified by the presence of the security icon present during an active call, as shown in the following example.

🔒

The presence of the security icon is the only visible indication that the media path is encrypted. The presence of this icon depends on whether the IP Deskphone has been configured to support SRTP or not and is visible when the IP Deskphone is not in the idle screen.



User2

4:27pm

To: Remote User
(remote@domain.com)
From: User2

Available SRTP configurations are provided in the following table.

Table 86: Configuration effects on media security display

Configuration	Result	Media Security Icon Display (during active call)
Default: UDP + TCP, no SRTP	SIP is unsecured; media is unsecured.	No
UDP + TCP. Best-Effort SRTP	SIP is unsecured; media is encrypted, but due to transmission of crypto parameters in clear text, the media cannot be considered secure.	No
UDP + TCP, SRTP-Only	SIP is unsecured; media is encrypted, but due to transmission of crypto parameters in clear text, the media cannot be considered secure.	No
TLS, no SRTP	SIP is secured; media is unencrypted.	No
TLS, Best-effort	SIP is unsecured; media is encrypted only if both end-points agree on use of SRTP.	Yes/No, depending on negotiation
TLS, SRTP Only	SIP is secured, media is encrypted. If both end-points do not agree on the use of SRTP, the connection fails.	Yes

The security icon indicates the security status of a call, and is useful for best-effort environments where there is a possibility of an unsecured call or where TLS is not used to communicate with the proxy.

Last successful or unsuccessful logon

You can configure the IP Deskphone to provide the user with logon feedback regarding the last successful logon or the last unsuccessful logon, and provide the local time at which logon feedback was logged (assuming that the IP Deskphone has the correct time configured). The time is correct when the IP Deskphone successfully retrieves the correct time during a successful logon process, or through the use of SNTP.

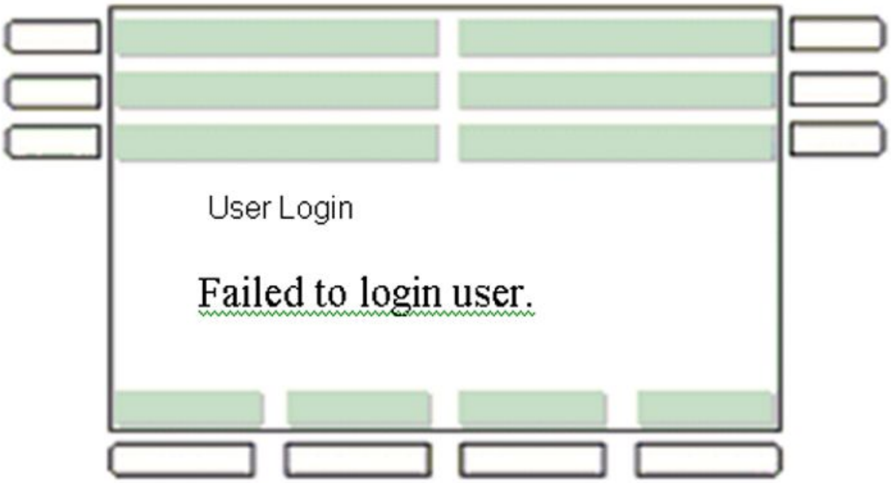
The display of a logon success and failure notification is local only to the IP Deskphone being used, and displays the last time that a user successfully logged on to the IP Deskphone or failed to log on to the IP Deskphone.

The figures shown below provide examples of the IP Deskphone display screen based on the configuration of the IP Deskphone and whether Login Notify is enabled or not.

The following notification appears on the display screen when the user login ID or password is incorrect and log in fails.

Note:

The server recognizes account login failure thresholds. After a configurable number of failures, the server temporarily disallows login attempts for an account. The IP Deskphone does not display any indication of this lockout.

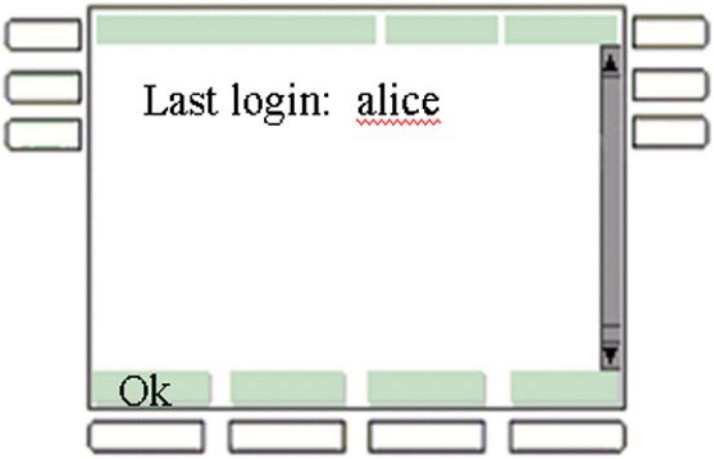


User Login

Failed to login user.

Figure 51: New login failure notification

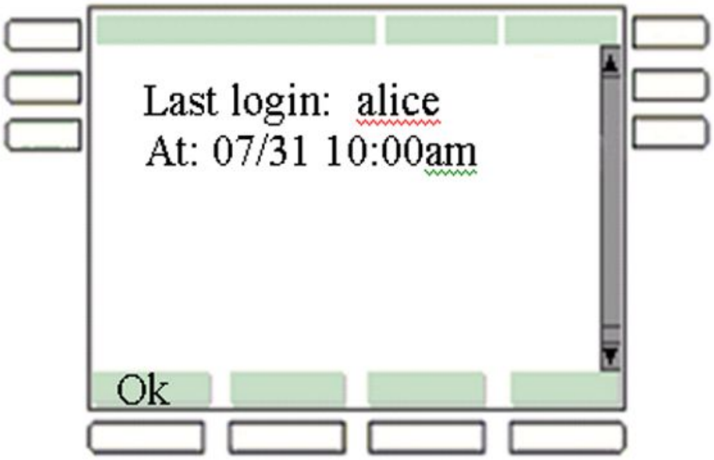
The following notification appears on the display screen when the user successfully logs on.



Last login: alice

Figure 52: Basic login notification

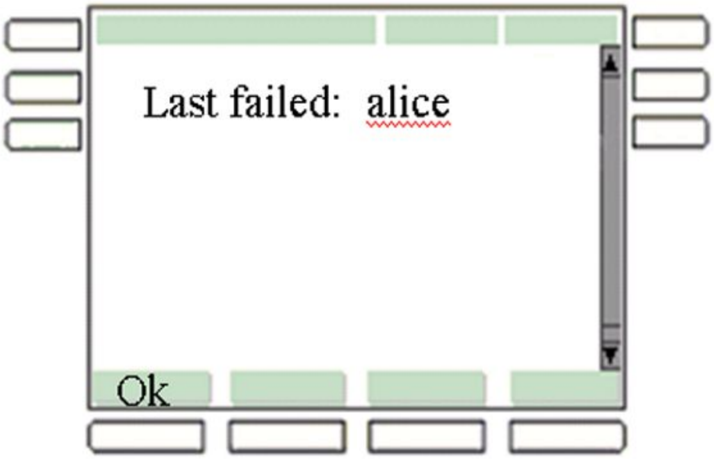
The following notification appears on the display screen when the user successfully logs on when Login Notify with Time is enabled.



Last login: alice
At: 07/31 10:00am

Figure 53: Basic login and time notification

The following notification appears on the display screen to notify the user of the last unsuccessful log on attempt made.



Last failed: alice

Figure 54: Login failure notification

The following notification appears on the display screen to notify the user of the date and time of the last unsuccessful log on attempt made.

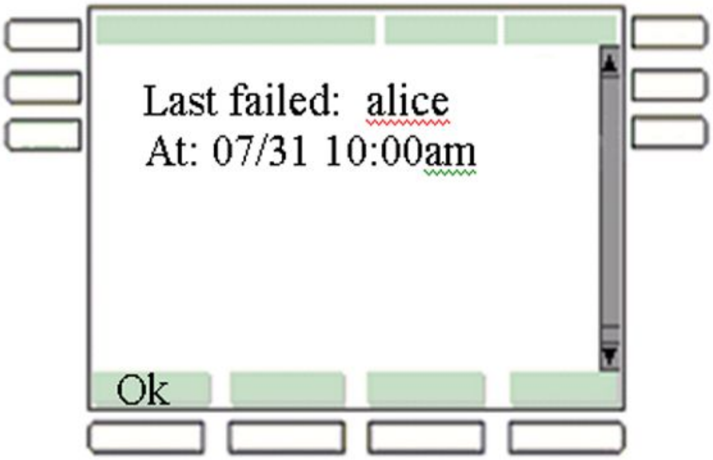
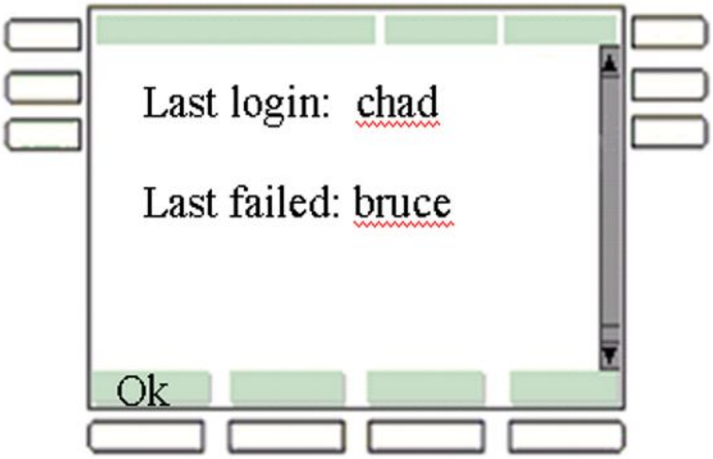

Last failed: alice
At: 07/31 10:00am

Figure 55: Login failure with time notification

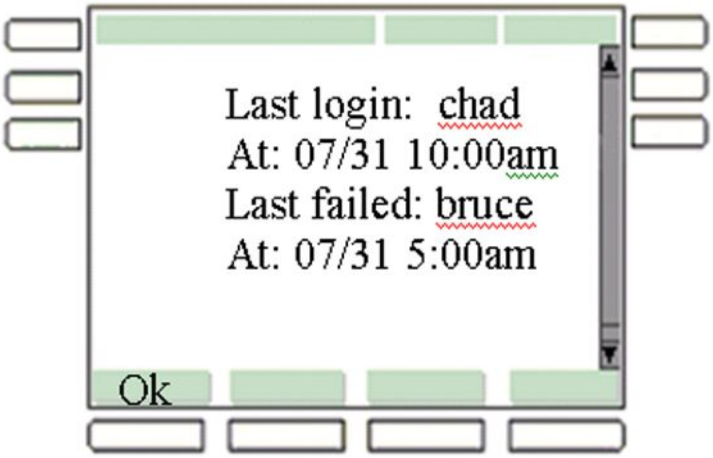
The following notification appears on the display screen to notify the user the last successful and unsuccessful log on attempts made.


Last login: chad

Last failed: bruce

Figure 56: Login and login failure notification

The following notification appears on the display screen to notify the user of the date and time of the last successful and unsuccessful log on attempts made.



Last login: chad
At: 07/31 10:00am
Last failed: bruce
At: 07/31 5:00am

Figure 57: Login and login failure with time notification

Enhanced administrative password security

The provisioning server can provide additional security associated with the administrative password. The provisioning server provides the password to the IP Deskphone in the form of an SHA1 or MD5 hash instead of the plain text password. This removes the need to store the password on the IP Deskphone by using the existing ADMIN_PASSWORD provisioning parameter.

The provisioning server can also enforce a password expiry using the provisioning flag, ADMIN_PASSWORD_EXPIRY. This flag contains a date after which the admin password stored on the IP Deskphone is not accepted. After this time, the administrative password must be changed in the administrative server. Password expiry can only be enforced if the date and time are retrieved by the IP Deskphone through SIP, SOAP, or SNTP.

Important:

IP Deskphone licensing information is located in the *Keycode Retrieval System (KRS) User Guide*. You must register for access to KRS.

Chapter 25: Licensing

A license is a "right to use" granted by Avaya, that the customer purchases to enable the features on the IP Deskphone.

The licensing framework uses an embedded server (node-locked or network locked) located in the IP Deskphone, to request tokens that allow the user to access any feature on the IP Deskphone.

Important IP Deskphone licensing information is located in the Keycode Retrieval System (KRS) User Guide. You must register for access to KRS.

Registering for access to KRS

1. Go to http://www.avaya.com/support.
2. Click **Online Self-Service**.
3. Select **Keycode Retrieval System**.
4. Select **GLOBAL LOGIN** from the list for the login location that you would like to use for access to the Keycode Retrieval System.
5. Select **IP CLIENTS** from the list for the product whose keycodes you would like to access.
6. When registration is validated, go to http://www.avaya.com/support and login in to KRS.
7. To view the KRS User Guide, select **Product family > Documentation > Forms and User Guides > KRS IP Clients User Guide_v2.ppt**.

Licensing framework

The licensing framework contains the fundamental infrastructure required to deliver a token-based licensing model that consists of a node-locked based licensing server and a licensing client.

The licensing framework consists of the following components:

- License Server (embedded)—Executes on the IP Deskphone and calls the server locally.
- License Client—Resides in the IP Deskphone and makes requests to the license server for tokens.
- KRS integration—A key or license generator provided with the CKLT solution, which is integrated into the Keycode Retrieval System (KRS).

The following figure shows a high-level view of the licensing framework interactions and components.

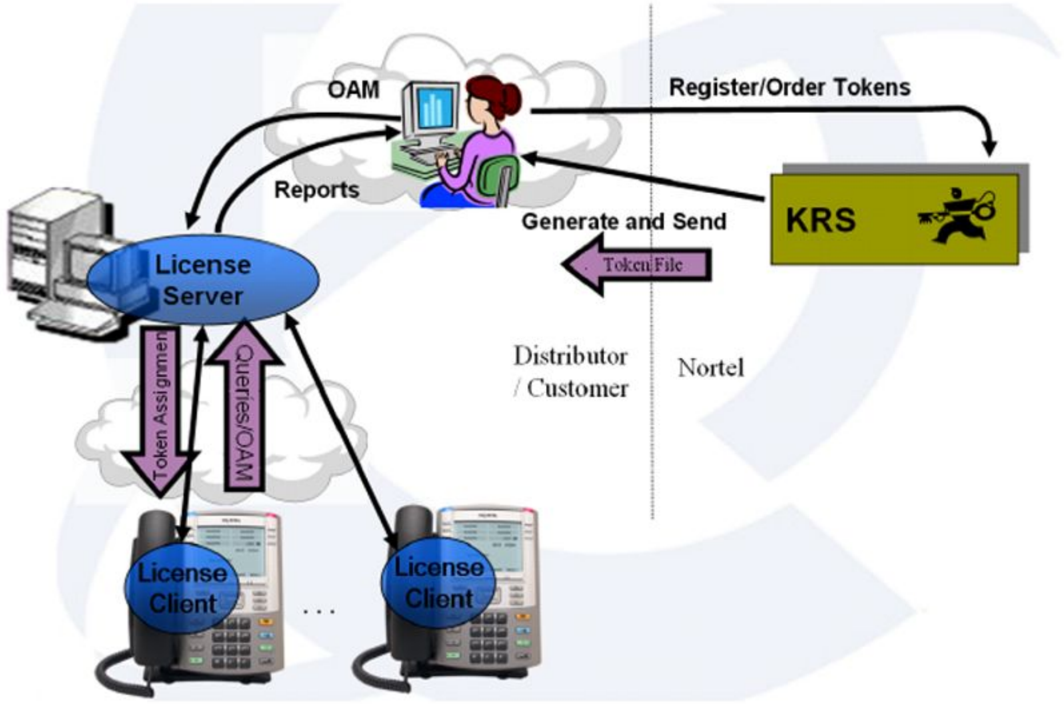


Figure 58: Licensing framework interactions and components

Characteristics of the licensing framework

The following list describes the characteristics of the licensing framework on the IP Deskphone.

- The embedded server (node-locked) enables the license server to execute on the IP Deskphone. The IP Deskphone obtains tokens by calling the server locally.
- The license file is installed on the IP Deskphone through the provisioning server or TFTP server.
- The IP Deskphone does not have a real-time clock. The time of day is obtained from the Call Server that the IP Deskphone is registered to on the network.
- The license file contains only one type of token because the IP Deskphone uses one type at a time.
- The administrator must enter the IP Deskphone system ID directly into the Keycode Retrieval System (KRS).
- The license file is keyed for the IP Deskphone so that the license is only valid on a specific IP Deskphone.
- The system ID is the MAC of the IP Deskphone.

- There is a one-to-one relationship between the license file and the IP Deskphone, therefore there are no multiple clients per server.
- The embedded server relies on a real time clock to calculate when a token expires.

Embedded server behavior

The embedded server does not provide the following capabilities:

- Grace period handling
- SSL communication with the IP Deskphone
- Crediting or transfer of entitlements
- Operation, Administration and Maintenance (OAM) capabilities

[LICENSING] section

The IP Deskphone config file must include a [LICENSING] section to enable the IP Deskphone to download the licence file. The [LICENSING] section specifies a wild card filename which uses the IP Deskphone MAC address as the filename with the cfg prefix and suffix.

The following format is an example of the [LICENSING] section that is added to the IP Deskphone config file (11xxe.cfg):

[LICENSING] VERSION version FILENAME X*.Y

The following table describes the items in the [LICENSING] section.

Table 87: Description of items in the [LICENSING] section of the config file.

Field name	Field value	Description
[LICENSING]	—	Section header for licensing config file information.
VERSION	000001	The version of the license file.
FILENAME	X*.Y	License filename. The IP Deskphone looks for a file with the IP Deskphone MAC address included in the filename.

The 11xxe.cfg file can have one, or all, of the following sections:

- [FW]
- [DEVICE_CONFIG]
- [LICENSING]

Although the IP Deskphone [FW] section is not required to activate the token, the provisioning server and the IP Deskphone provisioning server IP configuration must be configured to retrieve, save, and process the license file.

The following is an example of an 11xxe.cfg file that contains the [FW] section and the [LICENSING] section.

```
[FW] DOWNLOAD_MODE FORCED VERSION 0625C4E FILENAME 11xxes.bin
PROTOCOL TFTP SERVER_IP 47.11.183.165 SECURITY_MODE 0 [LICENSING]
VERSION 000001 FILENAME ipctoken*.cfg
```

The following is an example of an 11xxe.cfg file with the [LICENSING] section only.

```
[LICENSING] VERSION 000001 FILENAME ipctoken*.cfg
```

License file download

Use the following procedure to download the license file from the provisioning or TFTP server.

Downloading a license file

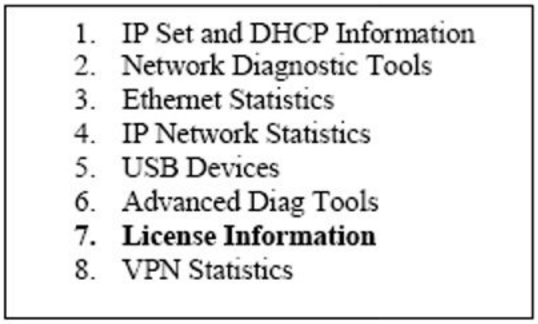
1. Configure the IP Deskphone with a provisioning IP address so it can access a provisioning server. For more information about provisioning parameters for the IP Deskphone, see Create the SIP provisioning file on the provisioning server.
2. Send the IP Deskphone license file to the provisioning server. The generated license file must be named iptokenMAC.cfg, where MAC is the 12-character MAC address of the IP Deskphone. For example, ipctoken000f1fd304f8.cfg.
3. Add [LICENSING] section to the IP Deskphone .cfg file, for example 1120eSIP.cfg, 1140eSIP.cfg, 1165eSIP.cfg, 1220eSIP.cfg, or 1230eSIP.cfg.

   For example: [FW] DOWNLOAD_MODE AUTO VERSION 3.00.xx.yy PROTOCOL TFTP FILENAME SIP 1140_e03.02.xx.yy.bin PROMPT NO [LICENSING’ DOWNLOAD_MODE_AUTO VERSION 000001 FILENAME ipctoken*cfg
4. Start the provisioning server so the IP Deskphone can retrieve the .cfg files when the server starts. Any new license file on the provisioning server overwrites the current file on the IP Deskphone. After the IP Deskphone retrieves the .cfg file during startup the IP Deskphone downloads the license file, renames the license file to ipclient.lic and saves the license file in the IP Deskphone Security File System (SFS).

License information for the IP Deskphone

To access the licensing feature, in the Diagnostics menu, choose License Information.

The following screen appears.



1. IP Set and DHCP Information
2. Network Diagnostic Tools
3. Ethernet Statistics
4. IP Network Statistics
5. USB Devices
6. Advanced Diag Tools
7. **License Information**
8. VPN Statistics

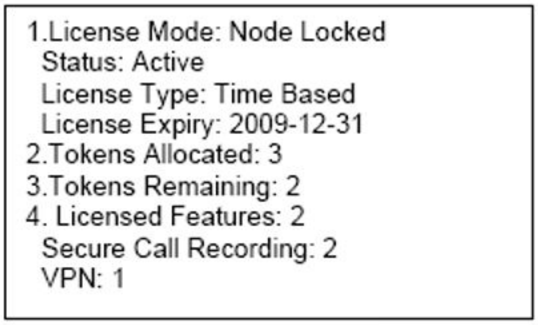
Figure 59: Local Diagnostics menu

Node-locked license mode

In the node-locked license mode, the IP Deskphone uses a license file to acquire the required tokens needed to activate the features. There are two types of tokens: time-based tokens, and SRS tokens.

Time-based token

The following figure is an example of a time-based token.



```
1.License Mode: Node Locked
  Status: Active
  License Type: Time Based
  License Expiry: 2009-12-31
2.Tokens Allocated: 3
3.Tokens Remaining: 2
4. Licensed Features: 2
  Secure Call Recording: 2
  VPN: 1
```

Figure 60: Node-locked license mode — license information for time-based token

The status of the time-based token can be one of the following:

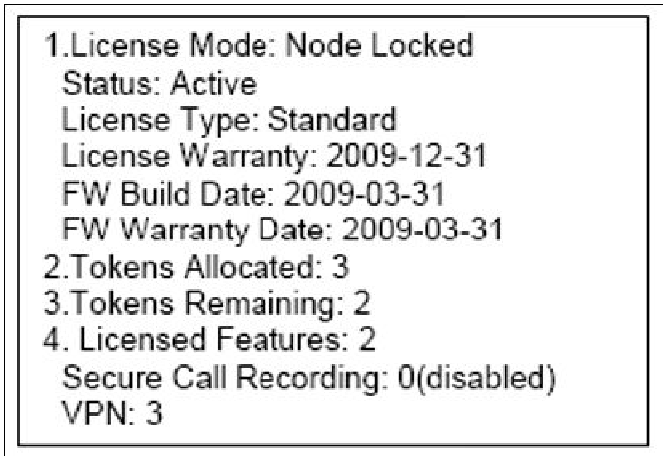
- Active
- Inactive

A time-based token can be Inactive for one of the following reasons:

- Insufficient token
- Licensed expired
- Released

Standard token

The following figure is an example of a standard token.



```
1.License Mode: Node Locked
  Status: Active
  License Type: Standard
  License Warranty: 2009-12-31
  FW Build Date: 2009-03-31
  FW Warranty Date: 2009-03-31
2.Tokens Allocated: 3
3.Tokens Remaining: 2
4. Licensed Features: 2
  Secure Call Recording: 0(disabled)
  VPN: 3
```

Figure 61: Node-locked license mode — license information for standard token

The status of the standard token can be one of the following:

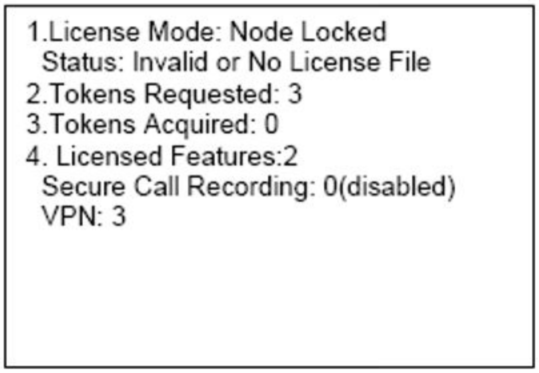
- Active
- Inactive

A standard token can be Inactive for one of the following reasons:

- Insufficient token
- Licensed expired
- Released

Invalid or no license file

The following figure is an example of an invalid or no license file.

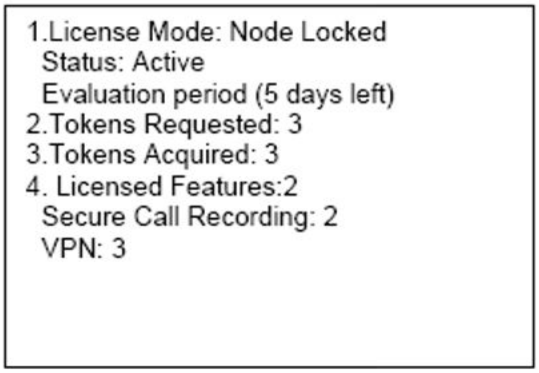


```
1.License Mode: Node Locked
  Status: Invalid or No License File
2.Tokens Requested: 3
3.Tokens Acquired: 0
4. Licensed Features:2
  Secure Call Recording: 0(disabled)
  VPN: 3
```

Figure 62: License information — Invalid or no license file

Evaluation period

The following figure is an example of the IP Deskphone in a 30 days evaluation period.



```
1.License Mode: Node Locked
  Status: Active
  Evaluation period (5 days left)
2.Tokens Requested: 3
3.Tokens Acquired: 3
4. Licensed Features:2
  Secure Call Recording: 2
  VPN: 3
```

Figure 63: License information — Evaluation period

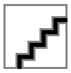
Network-locked license mode

In network locked license mode, the Avaya IP Deskphone does not require a license server to be configured. The IP Deskphone uses a generic license file in the IP Deskphone for required tokens to activate features. Network locked license modes provides the following types of tokens:

- Time based token
- Standard token

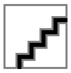
Time-based token

Time based tokens expire when the evaluation period ends or the license expires. The following figure shows the license information for a time-based token.



Standard token

The standard token is verified based on the software build and warranty dates. The following figure shows the license information for a standard token.



Alarms

The license feature provides notification on the IP Deskphone screen if the following conditions apply:

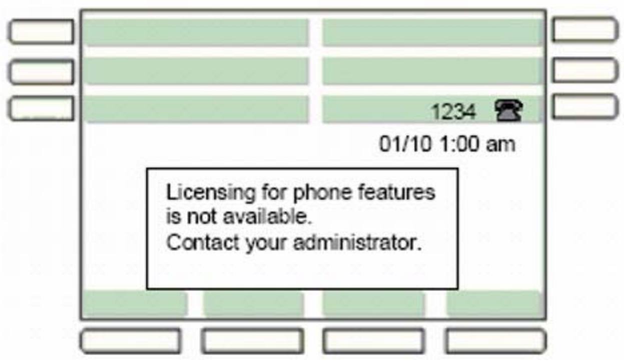
- tokens are not available
- tokens have expired
- evaluation period has ended

A notification message is displayed in a pop-up window on top of the "telephony" screen. The window can be dismissed by pressing the Stop key or by lifting the handset. After the message is dismissed, the IP Deskphone closes the warning window. The warning window re-displays every 24 hours at 1:00 am. You can configure the time frame through the IP Deskphone configuration system. If the licensed features are disabled, the IP Deskphone cannot display any type of window warning.

License not available warning

A warning window, indicating that a license is not available, appears on the IP Deskphone screen when the token request or refresh is rejected due to insufficient tokens available or an invalid license file.

The following figure is an example of a warning window indicating that a license is not available.



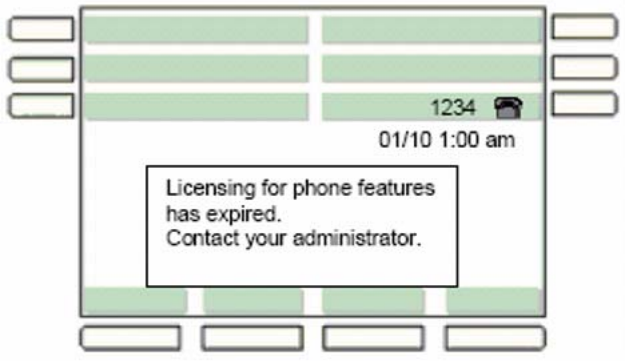
Licensing for phone features
is not available.
Contact your administrator.

Figure 64: License not available warning

License expiry warning

A warning window, indicating that a license has expired, appears on the IP Deskphone screen when a node-locked license expires.

The following figure is an example of a warning window indicating that a license is expired.



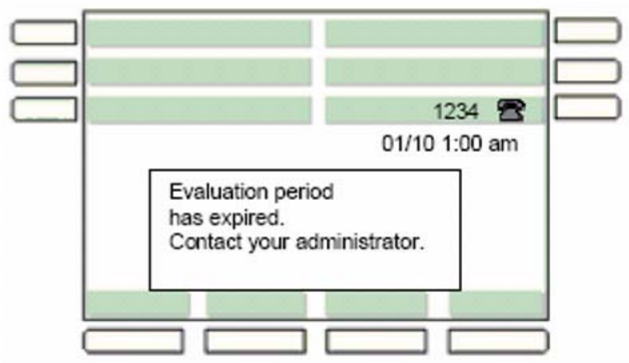
Licensing for phone features
has expired.
Contact your administrator.

Figure 65: License expiry warning

Evaluation period expiry warning

A warning window, indicating that the evaluation period has expired, appears on the IP Deskphone screen when the evaluation period expires and if the IP Deskphone has never had a valid token grant.

The following figure is an example of a warning window indicating that the evaluation license period is expired.



Evaluation period
has expired.
Contact your administrator.

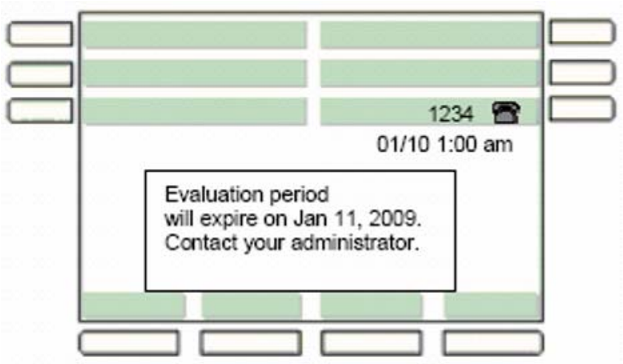
Figure 66: Evaluation period expiry warning

Evaluation threshold warning

A warning window informing you of the approaching evaluation expiration date appears on the IP Deskphone at the following predefined times:

- 15 days before expiration date
- 7 days before expiration date
- 1 day before expiration date

The following figure is an example of the evaluation threshold warning.



Evaluation period
will expire on Jan 11, 2009.
Contact your administrator.

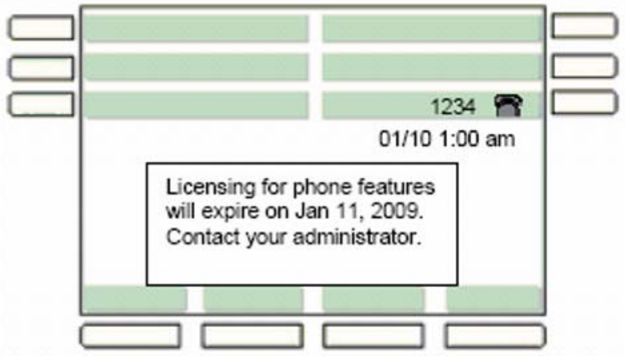
Figure 67: Evaluation threshold warning

Licensing threshold and grace period warning

When the expiration date for the grace period or the node-locked licence approaches, a warning window is displayed on the IP Deskphone. The warning window indicates when the license will expire, and notifies you at the following predefined times:

- 30 days before the license expires
- 15 days before the license expires
- 7 days before the license expires
- 1 day before the license expires

The following figure is an example of the license threshold and grace period warning.



Licensing for phone features
will expire on Jan 11, 2009.
Contact your administrator.

Figure 68: License threshold and grace period warning

Chapter 26: PC Client Softphone interworking

The interworking feature allows the user to access the functionality of the SIP IP Deskphone using a softphone client on their PC. On an incoming call, both the IP Deskphone and the PC Client Softphone ring. When the user answers the IP Deskphone, the softphone remains available for Instant Messages, video and other multimedia features.

The IP Deskphone, PC Client softphone, and the Call Server are all necessary to support interworking and the Click-to-Answer functionality.

The interworking feature enables the IP Deskphone to automatically answer an incoming call for the purpose of Click-to-Answer. To avoid any security risk, the user must pre-grant authorization to another user, or user groups, to allow them to make requests for the IP Deskphone to automatically answer their calls.

By using Click-to-Answer, the user can answer a call on their PC Client Softphone, causing the server to send an auto-answer request to the IP Deskphone. (When a user logs in, the IP Deskphone sends a special identifier so that only that specific IP Deskphone receives the request even though the user is logged in on multiple IP Deskphones.) The call is answered without user interaction, but the microphone is muted to prevent the device from being used as a listening device by a malicious user. When a call is answered, the user hears a ring-splash notification and can un-mute the microphone to allow bidirectional media.

Pre-granting authorization for the Answer-Mode

The user must specify which users or groups of users are authorized to request auto-answer. The user can grant authorization through the Feature Options menu if the interworking feature is enabled in the user's IP Deskphone device configuration.

The user can enable and disable one or more of the following groups:

- Allow Public—Authorizes anyone on the internet.
- Allow Friends List—Authorizes everyone on the user's Friends List.
- Allow Directory—Authorizes everyone in the user's Personal Directory.
- Allow Addresses—Acts as a white-list of domain names and SIP addresses that have authorized users.

Answer-Mode Settings screen

The Answer-Mode Settings screen is used to pre-grant authorization to request an automatic answer to potential callers or groups of callers.

The Answer-Mode Settings screen has the following two independent configurations:

- Allow Mode: [Current Setting]
- Allow Addresses

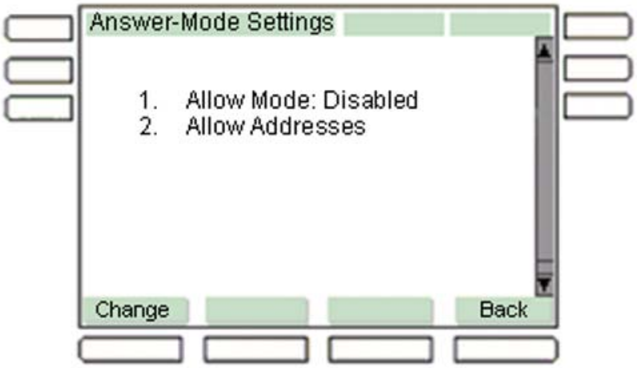
For the Allow Mode option, the current setting can be one of the following choices:

- Disabled
- Friends
- Directory—includes all Friends
- Public—includes all users

For the Allow Addresses option, the user can edit a listing by adding domain names or SIP addresses up to a maximum defined in the device configuration.

To access the Answer-Mode Settings screen, from the Preference menu, choose Feature Option and Answer-Mode Settings.

The following screen appears.



Answer-Mode Settings

1. Allow Mode: Disabled
2. Allow Addresses

Figure 69: Answer-Mode Settings screen

The following table describes the function of the Context-sensitive soft keys for the Answer-Mode Settings screen.

Table 88: Context-sensitive soft keys for the Answer-Mode Settings screen

Context-sensitive soft key	Action
Change	Opens the screen for the selected option: Allow Mode, or Allow Addresses.
Back	Returns you to the Feature Options screen.

The following table describes the outside actions on content for the Answer-Mode Settings screen.

Table 89: Outside actions on content for the Answer-Mode Settings screen

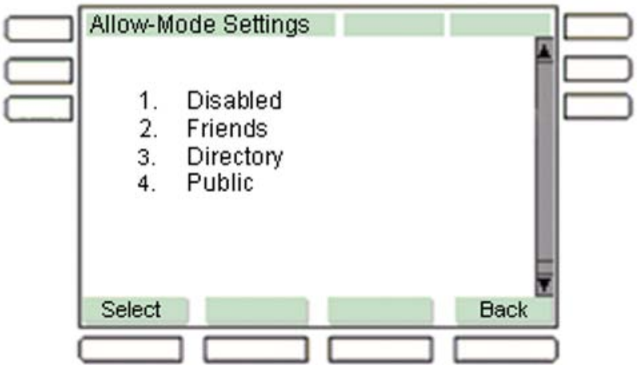
Key or action	Result
Goodbye	Idle screen.
Quit	Idle screen.
Off Hook; call keys	Clears the screen and allows you to make a call.

Allow-Mode Settings screen

The Allow-Mode Settings screen allows you to disable the feature, and to allow automatic requests for Friends, Directory, or Public users.

To access the Allow-Mode Settings screen, on the Preference menu, choose Feature Option, Answer-Mode Settings, and then Allow Mode.

The following screen appears.



Allow-Mode Settings

1. Disabled
2. Friends
3. Directory
4. Public

Figure 70: Allow-Mode Settings screen

The following table describes the function of the Context-sensitive soft keys for the Allow-Mode Settings screen.

Table 90: Context-sensitive soft keys for the Allow-Mode Settings screen

Context-sensitive soft key	Action
Select	Makes the current selection enabled. The selected user group is authorized for Answer-Mode.
Back	Returns you to the previous screen.

The following table describes the outside actions on content for Allow-Mode Settings screen.

Table 91: Outside action on content for the Allow-Mode Settings screen

Key or action	Result
Goodbye	Idle screen.
Quit	Idle screen.
Off Hook; call keys	Clears the screen and allows you to make a call.

Allow Addresses screen

The Allow Addresses screen is used to pre-grant authorization to request an automatic answer to a list of user-entered domains and SIP addresses.

If the user selects the Allow Addresses option in the Answer-Mode Settings screen, the user is presented with an interface for entering a list of strings. For the purpose of Click-to-Answer, only the current user is needed in the list because the requests originates from the user's PC Client Softphone.

For the Allow Addresses option, the user can edit a list of domain names or SIP addresses. The items in the list can be in any of the following formats:

- Single SIP user address

  For example:

  `sipuser@sipdomain.com`
- SIP domain

  For example:

  `sipdomain.com` (all users from `sipdomain.com`

- IPv4 address of a SIP domain

  For example:

  `172.25.20.20`
- IPv6 address of a SIP domain

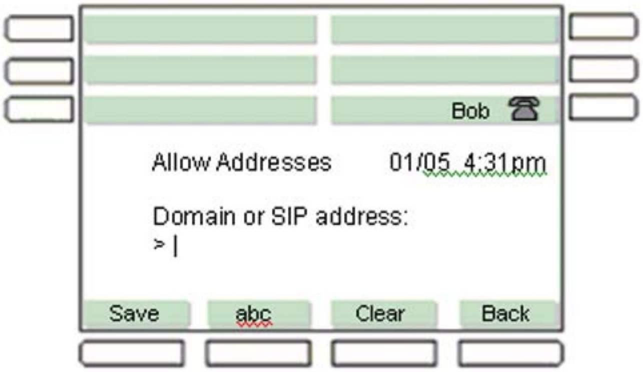
  For example:

  `2001:db8::57ab`

The user can add as many entries as the device configuration allows. If the Add soft key is disabled, then the user has reached the maximum number or entries. The user can also edit and delete entries.

To access the Allow Addresses screen, on the Preference menu, choose Feature Options, Answer-Mode Settings, and then Allow Addresses.

If there are no domains in the list, the following screen appears.

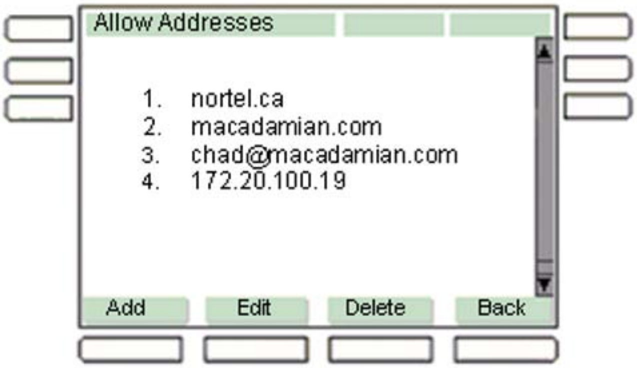


Allow Addresses 01/05 4:31pm

Domain or SIP address:
> |

Figure 71: Allow Addresses screen — first entry

The following screen is an example of the Allow Address screen if one (or more) domain or SIP address is in the system.



Allow Addresses

1. nortel.ca
2. macadamian.com
3. chad@macadamian.com
4. 172.20.100.19

Figure 72: Allow Addresses screen with domains and SIP addresses

The following table describes the function of the Context-sensitive soft keys for the Allow Addresses screen with no entries listed.

Table 92: Context-sensitive soft keys for the Allow Addresses screen - with no entries listed

Context-sensitive soft key	Action
Save	Saves the entered domain or SIP address in a list and displays the list content.
abc	Changes from alphanumeric and numeric entry (abc to 123).
Clear	Erases all entered characters.
Back	Returns you to the previous screen.

The following table describes the function of the Context-sensitive soft keys for the Allow Addresses screen with a list of entries.

Table 93: Context-sensitive soft keys for the Allow Addresses screen - with entries listed

Context-sensitive soft key	Action
Add	Displays the entry content.
Edit	Selects the current entry and displays the entry content with a populated field.
Delete	Deletes the selected domain from the list.
Back	Returns you to the Answer-Mode Settings screen.

The following table describes the outside actions on content for the Allow Addresses screens.

Table 94: Outside actions on content for the Allow Addresses screens

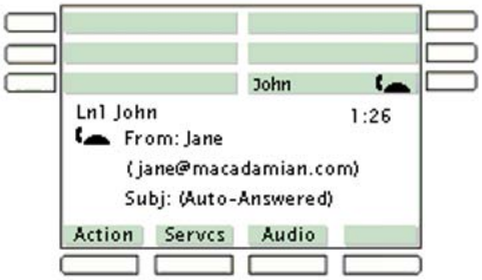
Key or action	Result
Goodbye	Idle screen.
Quit	Idle screen.
Off Hook; call keys	Clears the screen and allows you to make a call.

Automatically answering a call

With the interworking feature enabled, the IP Deskphone can answer automatically, manually, or reject an incoming auto-answer request. If the request is valid and the user is authorized to make the request (see Pre-granting authorization for the Answer-Mode on page 271), the call is answered automatically.

A "ring splash", or short ring tone, indicates to the user that the call was automatically answered. The subject is "Auto-Answered", and the microphone is muted (the user can deactivate the mute status by pressing the "mute" key on the IP Deskphone).

The following image is an example of a notification indicating an auto-answered call.



John ☎

Ln1 John 1:26

☎ From: Jane

(jane@macadamian.com)

Subj: (Auto-Answered)

Figure 73: Example of a Notification screen indicating an Auto-Answered call

When a call is auto-answered and the handset is on the hook, the handsfree button is activated.

If there is an active call when an auto-answer request is received, the active call is placed on hold and the incoming call is answered.

If a user who is not pre-granted authorization requests a call to be automatically answered on the IP Deskphone, the call is not automatically answered and is treated as a normal call; the IP Deskphone rings and the user answers it manually.

Configuration of the PC Client Softphone

Enabling the interworking feature in the IP Deskphone device configuration file allows the user to pre-grant authorization to other users and to configure the IP Deskphone to auto-answer.

The following table describes the configuration flags used to configure the PC Client Softphone interworking feature for the IP Deskphone.

Table 95: PC Client Softphone configuration commands

Configuration commands	Description
ENABLE_INTERWORKING	The configuration values are YES and NO. The default value is NO. • If configured as YES, the interworking feature is enabled. • If configured as NO, the interworking feature is disabled. If interworking is enabled, the interface for pre-authorization is visible in the Feature Options menu. If the feature is disabled, requests to automatically answer a call are handled as a normal incoming call. interworking must be configured through provisioning.
MAX_ALLOWEDADDRESSES	Limits the size of the list of user and domain addresses stored for auto-answered authorization. The default value is 100.

Chapter 27: Maintenance

Convert SIP Software to UNIStim Software

The IP Deskphone can be ordered with UNIStim software installed or with SIP Software installed. If you have an IP Deskphone with UNIStim software, and you convert the software from UNIStim to SIP, the UNIStim software is overwritten. To convert an IP Deskphone from SIP Software to UNIStim software, a software reload is required.

Reloading UNIStim software

1. Determine the appropriate UNIStim version to match the hardware release number of your IP Deskphone.

   There are different versions of UNIStim software available for download. Which version you choose depends on the hardware release number of your particular IP Deskphone.

   If the hardware release number of your IP Deskphone is among the following hardware release numbers, download UNIStim software version release 062AC5L or higher (the hardware release number is the Product Engineering Code [PEC] followed by the release number):

   - NTYS05ACE6 20
   - NTYS05BCE6 20
   - NTYS05BCGSE6 04

   If the hardware release number of your IP Deskphone is not among the previous list, download UNIStim version release 062AC5L or higher.
2. Download the appropriate UNIStim software file to your TFTP server.
3. Create an 12xxSIP.cfg file containing the following information:

   ```
   [FW]

   DOWNLOAD_MODE FORCED

   VERSION xxx

   FILENAME yyy.bin
   ```

where xxx is the UNIStim version number appropriate for the hardware release of your IP Deskphone, for example, 062AC5L, and SIP12x004.01.03.00.bin is the filename.

4. Power the IP Deskphone off and on. The IP Deskphone reboots and contacts the TFTP server upon bootup and downloads the new UNIStim software.

Reset Factory Settings support

A configured IP Deskphone can be reset to factory defaults to clear all stored information and preference data. By activating this mode, the data stored on the IP Deskphone is erased, and the administrator can reconfigure it for a new user.

The IP Deskphone resets data stored in the EEPROM to factory defaults and erases files in TFFS.

There are two ways to activate Reset to Factory Settings:

1. by entering a Special Key Sequence (SKS), or
2. remotely using SSH-PDT.

After you activate Reset to Factory Settings, the action is registered in the ECR-log file.

Activating Reset to Factory Setting by SKS

1. At any point while the IP Deskphone is operating, press the Special Key Sequence (SKS).
2. Enter the following command:

   `**73639<MAC>## (or **renew<MAC>##)`

   For example, the MAC-address, `A1B2C3D4E5F6` , can be translated to `212223343536` . Therefore, the SKS would be `**73639212223343536##` .

   After the proper sequence is entered on the IP Deskphone, the confirmation screen appears.
3. Press the Yes Context-sensitive soft key to reset to factory setting.

   Or

   Press the No Context-sensitive soft key to close the confirmation screen and return to regular mode.

The following table describes the function of the Context-sensitive soft keys for Reset to Factory Setting.

Table 96: Context-sensitive soft keys for Reset to Factory Setting

Context-sensitive soft key	Action
Yes	Activates Reset to Factory Setting.
No	Rejects Reset to Factory Setting, closes the confirmation screen and returns to regular mode.

Activating Reset to Factory Setting using SSH_PDT

1. Enter the PDT-command:

   ```
   >reset2factory
   ```

   The PDT displays the prompt:

   ```
   >Reset to Default... Are you sure?
   ```

2. Enter Y to accept.

   Or

   Enter N to decline.

   If you select Y, the PDT displays the prompt:

   ```
   >Enter MAC-address:
   ```

3. Type in the IP Deskphone MAC-address.

   ```
   ><MAC><enter>
   ```

   For example, if the IP Deskphone MAC-address is `A1B2C3D4E5F6` , you enter:

   ```
   >A1B2C3D4E5F6<enter>
   ```

4. Click Enter

   - If the MAC-address is correct, the IP Deskphone is reset and the remote telnet client is restarted.
   - If the MAC-address is incorrect, the IP Deskphone displays:

     ```
     >Incorrect MAC-address. Action is rejected .
     ```

     Return to Step 1.

SIP Software Web Manager

With the SIP Software Web Manager, the administrator can retrieve the backup of the configuration data (saved in the database) at any time or at the scheduled time. The backup archive can be downloaded to a PC and stored on the configured FTP server.

For more information on the SIP Software Web Manager, see Using the SIP Software Web Manager (NN43112-500)

Chapter 28: Diagnostics and troubleshooting

This chapter contains the following topics:

IP Deskphone diagnostics

Network-related issues can be debugged using the Network Diagnostic Utility (NDU) built into the IP Deskphone.

Another way to diagnose a problem on an IP Deskphone is to capture a message trace using any appropriate software.

The IP Deskphone has Problem Determination Tools (PDT). These can be accessed through a Telnet session using the IP address of the IP Deskphone (for login and password, contact Avaya).

Problem: Server unreachable after the IP Deskphone is powered up

If the display indicates that the server is unreachable and it continuously resets, some parameters must be configured. Things to consider when setting parameters:

- Enter requested information in the menu fields by pressing the number keys on the dialpad. Press the asterisk (*) key to enter a period (.) when entering an IP address.
- To record the entry and advance the initialization to the next parameter, press OK.
- To abandon the manual configuration process and restart the power-up, press Cancel.
- To manually enter parameters, use the BKSpace or Clear Context-sensitive soft keys to edit the default entry. BKSpace deletes each character as the key is pressed. Clear deletes the entire entry.
- Each parameter must have a corresponding entry. An audible beep indicates a field entry must be made before advancement to the next parameter.

Problem: Software Download Failure

If you are having trouble downloading software, review the following.

- Is the TFTP IP address correct within the IP Deskphone Device Settings menu?
- Is the IP Deskphone connecting to the TFTP server log?

  Check any firewall configuration settings to allow TFTP protocol access.

- Is the syntax within the 11xxe.cfg or 12xxSIP.cfg correct? See Configure the provisioning server on page 31. Supported sections describe the syntax of the configuration file.
- Does DOWNLOAD_MODE = AUTO and is VERSION less than the current running software version? If a file does not download using the AUTO selection, it is possible the version number is not high enough. A version number exists permanently on the IP Deskphone until a higher version number is downloaded through the device configuration file or you select **Srvcs, System, Erase User Data** on the IP Deskphone.
- Check FILENAME. Does this exist on the TFTP server?
- Check to make sure your firewall settings allow for the provisioning protocol (TFTP, FTP, OR HTTP) to go through.

There is a chance of software download failure, leaving the IP Deskphone with no valid application code and only the boot loaders. If this happens, the boot loaders execute and handle the application download.

Problem: Software Conversion Failures

There are four different boot loaders in the FLASH and application load. Various boot loaders are used to recover the IP Deskphone if a failure occurs.

- If a conversion fails before anything is written to FLASH, the IP Deskphone reboots with the UNIStim software load.
- If the software download fails while the application is being written to FLASH, there are two possible recovery methods:
  - If an application file was not created, after power up the IP Deskphone jumps to the BootC loader and downloads a new application load using the same mechanism as the application.
  - If the application is executed and the file created is corrupted, the IP Deskphone crashes. In this case, force the IP Deskphone to use BootC by pressing the UP key and “2” during power up.

Problem: Users of the IP Deskphone complain that their banner is not updated with their custom banner

When the banner is configured as FORCED in the device configuration file, the user's banner is overwritten by the value in the device configuration file.

Problem: Provisioning Error is displayed on the IP Deskphone display.

The Provisioning Error is displayed on the screen when the IP Deskphone is unable to contact the TFTP, FTP, or HTTP server.

Local diagnostic tools

Local diagnostic tools provides information about the IP Deskphone, such as identification, software version, settings, and a set of testing routines for checking network condition.

You can access Diagnostics tools through the Diagnostics menu.

Table 97: Diagnostics menu options on page 285, describes the Diagnostics menu options.

Table 97: Diagnostics menu options

Diagnostics option	Description
IP Deskphone and DHCP information	Provides detailed information about the IP Deskphone and service configuration.
Network Diagnostics Tools	Provides access to the following testing routines: • ping • tracert

Diagnostics option	Description
Ethernet Statistics	Provides some Ethernet statistics for Network Interface and PC port.
IP Network Statistics	Provides IP Network statistics.
Certificates Administration	Supports administration of available certificates.
Advanced Diag Tools	Provides information for setting up the following configuration parameters: • Auto Recovery (enable/disable) • SSH (enable/disable) • Port Mirroring (enable/disable) • User ID and Password for SSH
Test Key	Activates key testing mode.

How to access the Diagnostics menu

To activate the Diagnostics menu, access the Network menu by selecting one of the following steps:

- Press the Services key twice on the IP Deskphone while the IP Deskphone is in the idle mode.
- Press the Prefs Context-sensitive soft key, and then select the Network item in the Preferences menu.

The following screen appears:

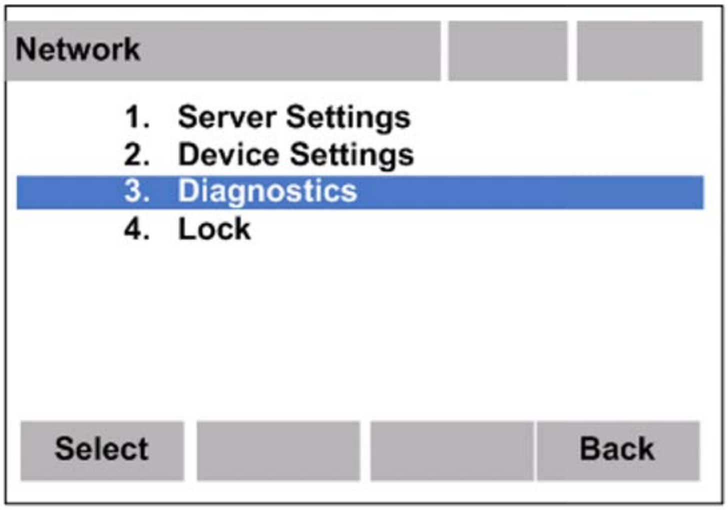


Figure 74: Network menu screen

After you access the Network menu, the following options are available:

- Server Settings
- Device Settings
- Diagnostics
- Lock

Select Diagnostics, or press Back to return to the Network menu.

The following screen appears:

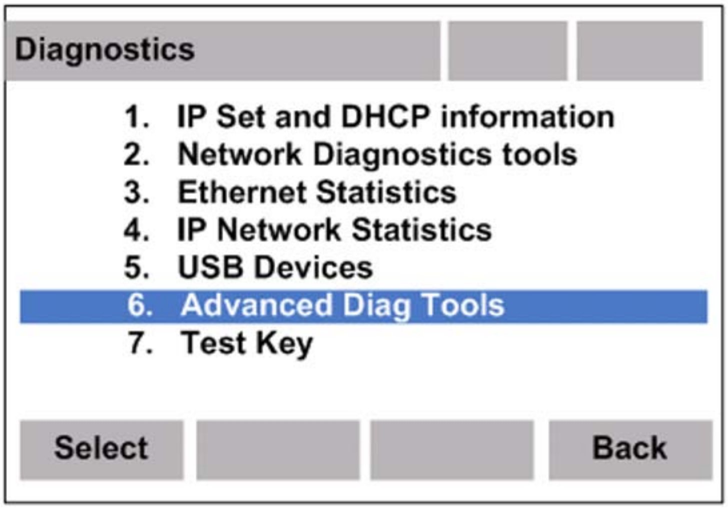


Figure 75: Diagnostics menu screen

The following table describes the function of the Navigation keys for the Diagnostics screen.

Table 98: Navigation

Key	Action
Up and down arrows	Use the up and down arrows to change the selected item in the list.
Enter	Invokes the Select Context-sensitive soft key.
Digital keys (number associated with option)	Invokes an appropriate option.
*	Selects the first option Server Settings, but does not activate it.
#	Selects the last option Lock, but does not activate it.

IP Set and DHCP information

The IP Set and DHCP information screen provides detailed information about the IP Deskphone, such as configuration, software version, IP addresses, gateway, and servers. To access the IP Set and DHCP information screen, from the Diagnostics menu, choose IP Set and DHCP information.

The following screen appears:

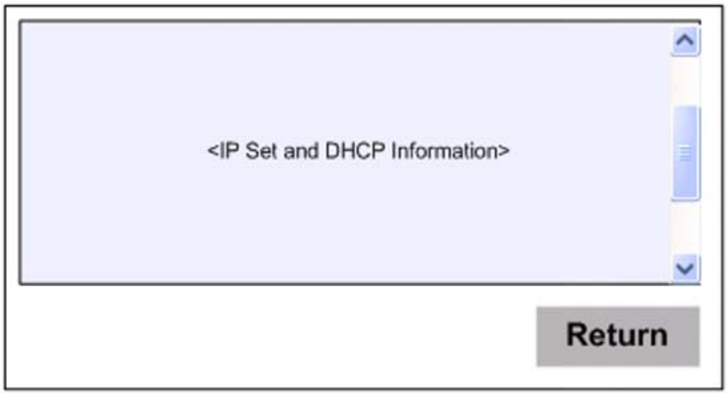


Figure 76: IP Set and DHCP information screen

The following is an example of the information that appears:

1. `Configuration Network Data Valid: Yes MAC Address Stored: Yes Perform DHCP: No Voice VLAN Enable: No Voice VLAN Config: No VLAN Voice VLAN Discovered: No`
2. `Primary Server: S1 PC Port is: ON`

3. `Software Version: 3.00.09.02 Hardware ID: xxxxxx`
4. `Set IP: xxx.xxx.xxx.xxx (could be in IPv4 or IPv6 format)`
5. `Sub-Mask: xxx.xxx.xxx.xxx (could be in IPv4 or IPv6 format)`
6. `GateWay: xxx.xxx.xxx.xxx (could be in IPv4 or IPv6 format)`
7. `Voice VLAN Priority: 6`
8. `Voice VLAN ID: 6`
9. `DHCP Respond String: ....`
10. `Servers' Information: S01 IP: xxx.xxx.xxx.xxx Port: 4100 Act: 1 Retries: 5 S02 IP: xxx.xxx.xxx.xxx Port: 4100 Act: 1 Retries: 5 S03: IP: xxx.xxx.xxx.xxx Port: 4100 Act: 1 Retries: 5 S04 IP: xxx.xxx.xxx.xxx Port: 4100 Act: 1 Retries: 5`
11. `Provisioning Server: xxx.xxx.xxx.xxx`

The following table describes the function of the Context-sensitive soft keys for the IP Set and DHCP Information screen.

Table 99: Context-sensitive soft key for the IP Set and DHCP information screen

Context-sensitive soft key	Action
Up and down arrows	Use the up and down arrows to scroll the screen.

The following table describes the function of the Navigation key for the IP Set and DHCP Information screen.

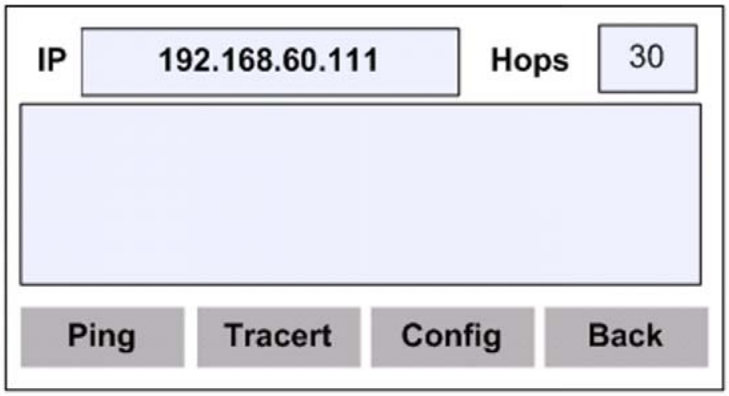
Table 100: Navigation

Key	Action
Return	Press the Return Context-sensitive soft key to cancel this screen and return to the Diagnostics menu.

Network Diagnostics tools

The Network Diagnostics tools menu provides access to ping and tracert testing routines. To access the Network Diagnostics tools screen, from the Diagnostics menu, choose Network Diagnostics tools.

The following screen appears:



IP 192.168.60.111 Hops 30

Figure 77: Network Diagnostics tools screen

The screen contains two configurable fields:

- IP—enter an IP address.
- Hops—number of hops used as a configurable parameter for tracert routine.

The following services are available:

- activate the ping routine
- activate the tracert routine
- activate a dialog for additional configurable parameters

The following table describes the function of the Context-sensitive soft keys for the Network Diagnostics tools screen.

Table 101: Context-sensitive soft keys for the Network Diagnostics tools screen

Context-sensitive soft key	Action
Ping	Activates the ping routine.
Tracert	Activates the tracert routine.
Config	Activates a menu for additional configurable parameters.
Back	Returns you to the Diagnostics menu.

The following table describes the function of the Navigation keys for the Network Diagnostics tools screen.

Table 102: Navigation

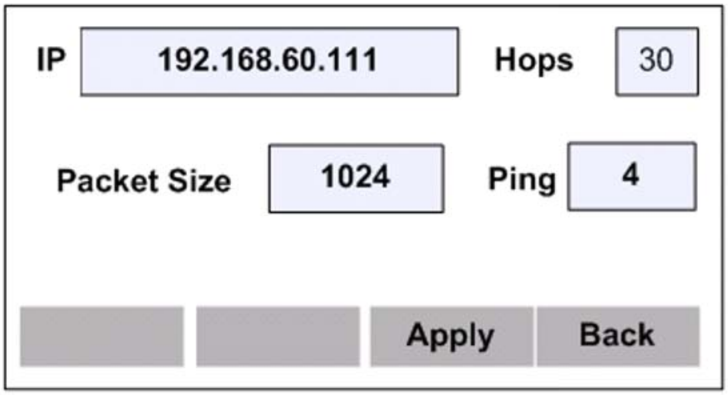
Key	Action
Up and down arrows	Use the up and down arrows to scroll through a list of testing information.

Key	Action
Left and right arrows	Use the left and right arrows to move through the configurable fields.
Enter	Use the Enter key to enter the editing mode for the active configurable field.

Config option in Network Diagnostics tools

The Config screen provides access to additional configurable parameters used by testing routines. You can access the screen by pressing the Config Context-sensitive soft key on the IP Deskphone after the Network Diagnostics tools screen is active.

The following screen appears:



IP 192.168.60.111 Hops 30

Packet Size 1024 Ping 4

Figure 78: Network Diagnostics tools (Config) screen

The screen contains the following configurable fields:

1. IP—The user can enter an IP address.
2. Hops—The number of hops used as a configurable parameter for tracert routine.
3. Packet Size—Size of the network packet used by the ping routine.
4. Ping—The number of ping packages.

The following table describes the function of the Context-sensitive soft keys for the Network Diagnostics tools (Config) screen.

Table 103: Context-sensitive soft keys for the Network Diagnostics (Config) screen

Context-sensitive soft key	Action
Apply	Applies settings, dismisses the screen, and returns you to the Network Diagnostics menu.

Context-sensitive soft key	Action
Back	Returns you to the Diagnostics menu.

The following table describes the function of the Navigation keys for the Network Diagnostics tools (Config) screen.

Table 104: Navigation

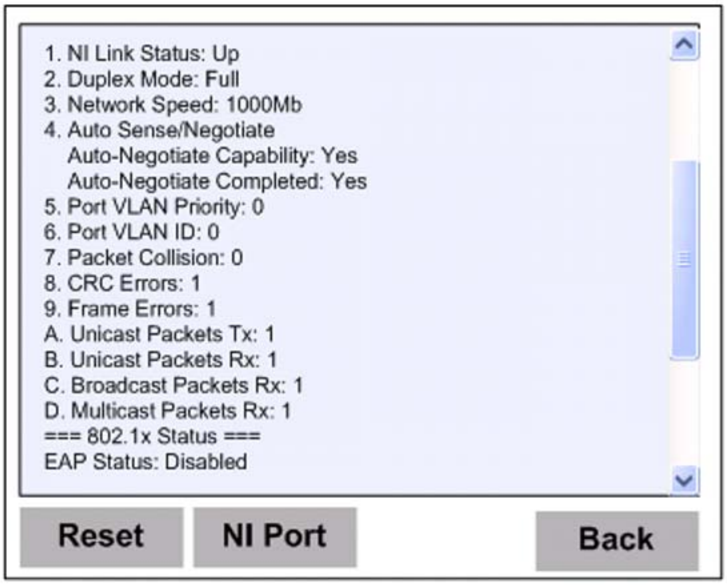
Key	Action
Left and right arrows	Use the left and right arrows to move through the configurable fields.
Enter	Use the Enter key to enter the editing mode for the active configurable field

Ethernet Statistics

The Ethernet Statistics (NI Port) screen displays ethernet statistics information for Network Interface (NI) or PC ports, such as the number of incoming and outgoing network packages and network settings.

To access the Ethernet Statistics screen, from the Diagnostics menu, choose Ethernet Statistics.

The following screen appears:



1. NI Link Status: Up
2. Duplex Mode: Full
3. Network Speed: 1000Mb
4. Auto Sense/Negotiate
   Auto-Negotiate Capability: Yes
   Auto-Negotiate Completed: Yes
5. Port VLAN Priority: 0
6. Port VLAN ID: 0
7. Packet Collision: 0
8. CRC Errors: 1
9. Frame Errors: 1
A. Unicast Packets Tx: 1
B. Unicast Packets Rx: 1
C. Broadcast Packets Rx: 1
D. Multicast Packets Rx: 1
=== 802.1x Status ===
EAP Status: Disabled

Figure 79: Ethernet Statistics (NI Port) screen

The following is an example of Ethernet Statistics for the IP Deskphone:

```
1. NI Link Status: Up 2. Duplex Mode: Full 3. Network Speed: 1000Mb
4. Auto Sense/Negotiate Auto-Negotiate Capability: Yes Auto-Negotiate
Completed: Yes 5. Port VLAN Priority: 0 6. Port VLAN ID: 0 7. Packet
Collision: 0 8. CRC Errors: 1 9. Frame Errors: 1 A. Unicast Packets
Tx: 1 B. Unicast Packets Rx: 1 C. Broadcast Packets Rx: 1 D. Multicast
Packets Rx: 1 === 802.1x Status === EAP Status: Disabled
```

The following table describes the function of the Context-sensitive soft keys for the Ethernet Statistics (NI Port) screen.

Table 105: Context-sensitive soft keys for the Ethernet Statistics (NI Port) screen

Context-sensitive soft key	Action
Reset	Resets statistics value.
NI Port	Switches to the PC Port Ethernet statistics.
Back	Returns you to the Diagnostics menu.

The following table describes the function of the Navigation keys for the Ethernet Statistics (NI Port) screen.

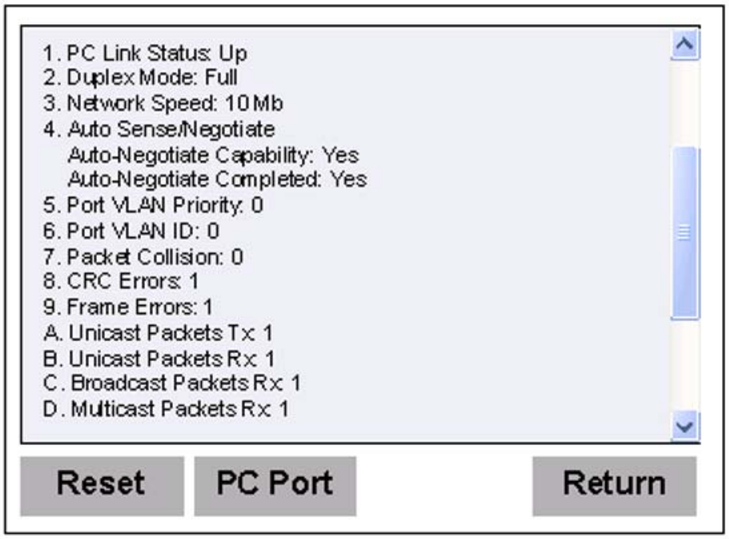
Table 106: Navigation

Key	Action
Up and down arrows	Use the up and down arrows to scroll through a list of statistics information.

Ethernet Statistics (PC Port) screen

The Ethernet Statistics (PC Port) screen displays ethernet statistics for the PC port. To access the PC Port, from the Ethernet Statistics screen, press the NI Port Context-sensitive soft key.

The following screen appears:



1. PC Link Status: Up
2. Duplex Mode: Full
3. Network Speed: 10 Mb
4. Auto Sense/Negotiate
   Auto-Negotiate Capability: Yes
   Auto-Negotiate Completed: Yes
5. Port VLAN Priority: 0
6. Port VLAN ID: 0
7. Packet Collision: 0
8. CRC Errors: 1
9. Frame Errors: 1
A. Unicast Packets Tx 1
B. Unicast Packets Rx 1
C. Broadcast Packets Rx 1
D. Multicast Packets Rx 1

Figure 80: Ethernet Statistics (PC Port) screen

The following is an example of Ethernet Statistics for the PC Port:

```
1. PC Link Status: Up 2. Duplex Mode: Full 3. Network Speed: 10 Mb 4.
Auto Sense/Negotiate Auto-Negotiate Capability: Yes Auto-Negotiate
Completed: Yes 5. Port VLAN Priority: 0 6. Port VLAN ID: 0 7. Packet
Collision: 0 8. CRC Errors: 1 9. Frame Errors: 1 A. Unicast Packets
Tx: 1 B. Unicast Packets Rx: 1 C. Broadcast Packets Rx: 1 D. Multicast
Packets Rx: 1
```

The following table describes the function of the Context-sensitive soft keys for the Ethernet Statistics (PC Port) screen.

Table 107: Context-sensitive soft keys for the Ethernet Statistics (PC Port) screen

Context-sensitive soft key	Action
Reset	Resets statistics values.
PC Port	Switches to the NI Port Ethernet statistics.
Back	Returns you to the Diagnostics menu.

The following table describes the function of the Navigation keys for the Ethernet Statistics (PC Port) screen.

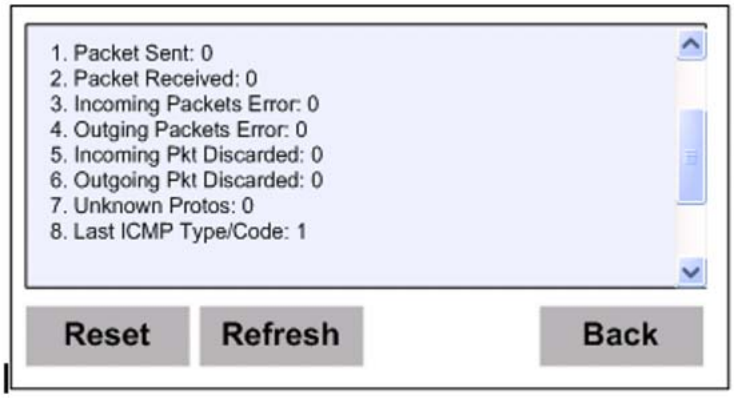
Table 108: Navigation

Key	Action
Up and down arrows	Use the up and down arrows to scroll through a list of statistics information.

IP Network Statistics

The IP Network Statistics screen provides information such as the number of incoming and outgoing network packages, number of error packages, and protocols. To access the IP Network Statistics screen, from the Diagnostics menu, choose IP Network Statistics.

The following screen appears:



1. Packet Sent: 0
2. Packet Received: 0
3. Incoming Packets Error: 0
4. Outging Packets Error: 0
5. Incoming Pkt Discarded: 0
6. Outgoing Pkt Discarded: 0
7. Unknown Protos: 0
8. Last ICMP Type/Code: 1

Figure 81: IP Network Statistics screen

The following is an example of IP Network Statistics for the IP Deskphone:

```
1. Packet Sent: 0 2. Packet Received: 0 3. Incoming Packets Error: 0
4. Outgoing Packets Error: 0 5. Incoming Pkt Discarded: 0 6. Outgoing
Pkt Discarded: 0 7. Unknown Protos: 0 8. Last ICMP Type/Code: 1
```

The following table describes the function of the Context-sensitive soft keys for the IP Network Statistics screen.

Table 109: Context-sensitive soft keys for the IP Network Statistics screen

Context-sensitive soft key	Action
Reset	Resets statistics values.
Refresh	Refreshes the IP Network statistics.

Context-sensitive soft key	Action
Back	Returns to the Diagnostics menu.

The following table describes the function of the Navigation keys for the IP Network Statistics screen.

Table 110: Navigation

Key	Action
Up and down arrows	Use the up and down arrows to scroll through a list of statistics information.

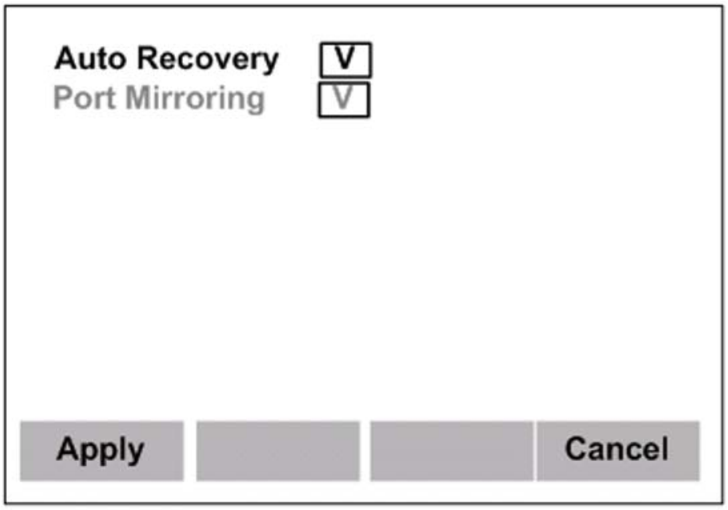
Advanced Diag Tools

With the Advanced Diag Tools option, you can modify the following parameters:

- Auto Recovery (enable/disable)
- Port Mirroring (enable/disable)
- SSH-SFTP (enable/disable)
- User Id and Password for SSH

To access the Advanced Diag Tools screen, from the Diagnostics menu, choose Advanced Diag Tools.

The following screen appears:



Auto Recovery ☑

Port Mirroring ☑

Figure 82: Advanced Diag Tools screen

The following table describes the function of the context-sensitive soft keys for the Advanced Diag Tools screen.

Table 111: Context-sensitive soft keys for the Advanced Diag Tools screen

Context-sensitive soft key	Action
Apply	Invokes the selected service.
Cancel	Dismisses the dialog box and returns you to the Diagnostics menu.

The following table describes the function of the navigation keys for the Advanced Diag Tools screen.

Table 112: Navigation

Key	Action
Up and down arrows	Use the up and down arrows to scroll through a list of statistics information.
Enter	Use the Enter key to enter the editing mode for the active configurable field or change the value for check boxes.

Port Mirroring

The Port Mirroring field behavior depends on the device configuration flag defined by the administrator. A device configuration file parameter manages the PC Port Mirroring option:

PORT_MIRROR_ENABLE Yes/[No]

The command determines whether or not the option can be managed:

- If PORT_MIRROR_ENABLE is Yes, then you can activate or deactivate the option. The Port Mirroring prompt in the Advanced Diag Tools menu is enabled and can be modified.
- If PORT_MIRROR_ENABLE is No, then you cannot manage the option. The Port Mirroring prompt in the Advanced Diag Tools menu is disabled (dimmed); Port Mirroring is disabled.

The default value for the PORT_MIRROR_ENABLE is No. This means that PC Port Mirroring is not active.

SSH-SFTP

For information about enabling SSH-SFTP, see “Manually enabling SSH and SFTP” (page 286).

Author's note: insert link in the line above.

User ID and Password for SSH

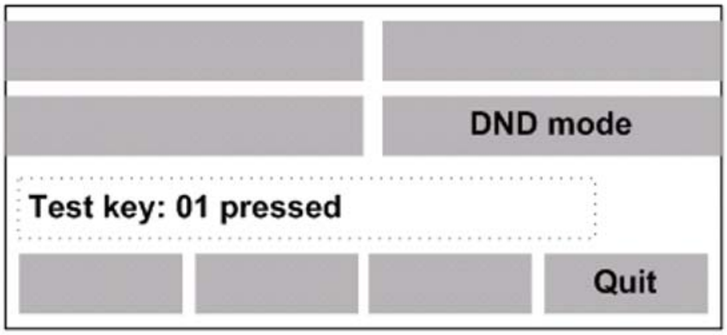
For information about the User ID and Password for SSH, see “Manually enabling SSH and SFTP” (page 286).

Author's note: insert link in the line above.

Test key

The Test key screen lets you perform a physical key operation test. After you activate the test mode, the "Test key: Press any key" prompt appears on the screen. The IP Deskphone goes into the Do Not Disturb (DND) mode and cannot receive any external calls. Information about the pressed key event (except for the Rls key) appears on the IP Deskphone screen. To access the Test key screen, from the Diagnostics menu, choose Test key.

The following screen appears:



Test key: 01 pressed

Figure 83: Test key screen

After you activate the test mode, the key event appears on the screen:

- `Key pressing: "Test key: xx pressed"`
- `Key pressing: "Test key: xx pressed"`

The following table describes the function of the Context-sensitive soft keys for the Test key screen.

Table 113: Context-sensitive soft keys for the Test key screen

Context-sensitive soft key	Action
Quit	Dismisses the Services menu.

The following table describes the function of the Navigation key for the Test key screen.

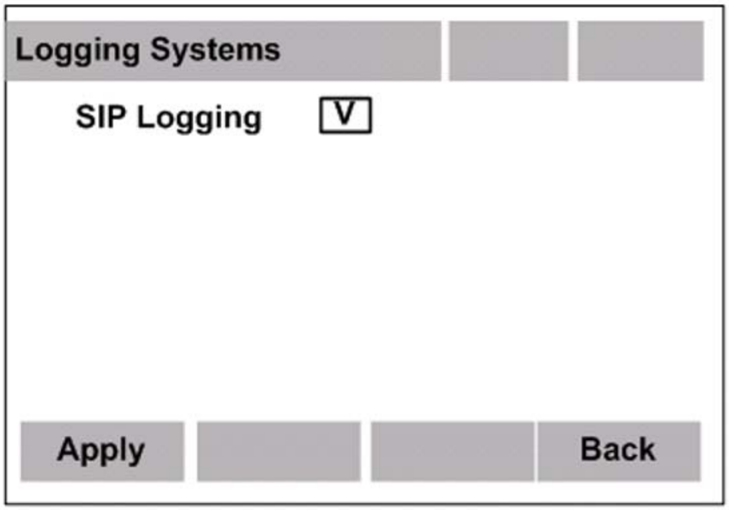
Table 114: Navigation

Key	Action
Rls	Closes the test mode and restarts the IP Deskphone.

Logging System

Logging System contains a subsystem for logging incoming and outgoing SIP packages to the log file in FFS, for the IP Deskphone. You can enable or disable the SIP logging subsystem by selecting the check box for ON (enable) or deselecting the check box for OFF (disable). To access the Logging System menu, press the Services key on the IP Deskphone, and then choose Logging System from the Services menu.

The following screen appears:



Logging Systems

SIP Logging V

Figure 84: Logging Systems screen

The Logging Systems screen displays the SIP Logging subsystem. Press the Enter key in the Navigation key cluster to switch the value of the selected sign from ON to OFF, or OFF to ON. Then press the Apply Context-sensitive soft key to apply the settings.

The following table describes the function of the Context-sensitive soft keys for the Logging Systems screen.

Table 115: Context-sensitive soft keys for the Logging Systems screen

Context-sensitive soft key	Action
Apply	Applies the setting and returns to the parent screen.
Back	Dismisses the setting and returns you to the parent screen.

The following table describes the function of the Navigation keys for the Logging Systems screen.

Table 116: Navigation

Key	Action
Up and down arrows	Use the up and down arrows to scroll the screen.
Right and left arrows	Navigates through the signs.
Enter	Switches the value of the selected sign from ON to OFF, and OFF to ON.

You can enable of disable SIP-logging using the following command in the Device configuration file:

LOGSIP_ENABLE Yes/[No]

If the parameter is Yes, the SIP-logging Manager is active and starts logging SIP incoming and outgoing packages into the log file in FFS. If the parameter is No, the SIP-logging Manager is not active and there is no logging of incoming and outgoing packages into the log file in FFS. The Default parameter is No.

Problem Determination Tool (PDT)

The IP Deskphone with SIP Software contains special services that monitor the performance and various other states of the IP Deskphone. These services also automatically collect problem data, and provide symptom analysis support for the various categories of problems encountered by the software. All significant events are registered in special log files.

Error Logging framework

The Error logging framework saves error-related information in the ECR Log file and is the base object used by all the other monitoring services listed as follows:

- ECR Watchdog
- Task Monitor
- CPU Load Monitor
- Stack Overflow Monitor
- Traffic Monitor

ECR Watchdog

The ECR Watchdog tracks the IP Deskphone to ensure the IP Deskphone survives transitions (for example: soft reset). If the watchdog is active and has not detected activity in a certain period of time, the watchdog logs the appropriate error and recovers the IP Deskphone.

Task Monitor

The Task Monitor performs the following functions:

- Tracks the switch of any task to the suspended state. If the task gets to the suspended state, the Task Monitor logs the error-related information (including the suspended task information and summary information about all running tasks), and then initiates recovery of the IP Deskphone.

- Monitors important tasks. The Task Monitor scans these tasks, and if any task is lost without a reason, the Task Monitor logs the error and recovers the IP Deskphone.

CPU Load Monitor

The CPU Load Monitor tracks the CPU usage. If the CPU load reaches 100 percent and stays at that level for more than 1 minute, The CPU Load Monitor logs the appropriate error (including the list of most suspect tasks that could occupy the CPU), and recovers the IP Deskphone.

Stack Overflow Monitor

The Stack Overflow Monitor tracks the stack of all tasks in the real-time mode, detects the stack overflow or corruption, and logs the task trace.

Traffic Monitor

The Traffic Monitor monitors incoming and outgoing IP and SIP traffic and registers events in the ECR-log file when the traffic exceeds predefined thresholds. The Traffic Monitor also registers the content of the incoming and outgoing SIP packages.

The PDT commands

The PDT is a troubleshooting tool for the IP Deskphone. The PDT has powerful functions which allow you to perform special testing actions, and can display the content of any log files. The PDT helps to identify the origin of the problem under investigation, reduces the amount of time it takes to reproduce a problem with the proper RAS tracing levels set (trace levels are set automatically by the tool), and reduces the effort required to send the appropriate log information to technical support.

The PDT provides remote access to the IP Deskphone with the problem, using a Telnet session. Access is restricted by admin ID and password.

The PDT supports the following set of commands:

Table 117: List of PDT commands

#	Command	Description
1	prtlog >prtlog<mngr_dest>	• Prints a content of the ECR-log file. • Outputs content of the specified log file to stdout (the screen, a stream, stdout, or a string). The input parameter specifies a type of logging manager: - 0 (default)—ECR-log file - 1—SIP-log file If the input parameter is incorrect, the following notification appears: >prtlog: incorrect type of manager <x>
2	clearLogFile	Clears a content of the ECR-log file
3	setLogLevel <loglevel>	Configures log level, where the loglevel is in the range 0...3: • If loglevel == 0—logging disabled • If loglevel == 1—logging only Critical errors • If loglevel == 2—logging Critical and Major errors • If loglevel >=3—logging any type of errors
4	printLogLevel	Print log level
5	setRecoveryLevel <reclevel>	Sets up recovery level, where the reclevel is in the range 0...3. If the Auto Recovery option is ON, the IP Deskphone behaves as follows: • If reclevel == 0—recovering disabled • If reclevel == 1—recovering on only Critical errors • If reclevel == 2—recovering on Critical and Major errors • If reclevel >= 3—recovering on any errors
6	printRecoveryLevel	Prints recovery level
7	taskMonShow	Prints a list of monitored tasks
8	"i"	Prints all task information
9	ti <taskName | task id>	Print task information
10	memshow [level]	Show memory information

#	Command	Description
11	checkStack <taskName | task id>	Check stack of some task
12	tt <taskName | task id>	Print Task Trace
13	info	Print HardwareID, SoftwareID, MAC and BT address
14	prtcfg	Prints content of the IP Deskphone configuration file, SystemConfig.dat in FFS. The file contains IP Deskphone-specific configuration. The content of this file is formed from the content of several downloadable configuration files: • Device Configuration file • Tones file • Language file
15	lsr	List directory contents (similar to unix ls) and the contents of a directory and any of its subdirectories
16	ping <host ip> [# of pings]	Ping any host (ping)
17	tracert <host ip> [max hops]	Traceroute to any host (tracert)
18	netinfo	Print common network information
19	routeshow	Display host and network routing tables and stats
20	arp	Display entries in the system ARP table
21	listcerts	List all trusted certificates
22	printcert <index>	Display certificate details
23	sipapp <start | stop>	Start or stop the SIP application
24	sendunistim <xx xx....>	Send UNIStim message
25	rxunistim <on | off	Display UNIStim messages from the Core
26	txunistim <on | off	Display UNIStim messages to the Core from the SIP application
27	sendevent <0xmmm <0xnnn>	Simulate an UNIStim event.
28	lcdparam	Set up LCD parameters for the IP Deskphone
29	audio <hs | hd | hf | off>	Loopback audio to handset/headset/Handsfree
30	display <on | off>	Turn all LCD and LED on or off
31	keypress <on | off>	Turn all key presses on or off

#	Command	Description
32	clearlog >clearlog<mngr_dest>	Clears content of the specified log file. The input parameter specifies the type of logging manager: • 0 (default)—ECT-log file • 1—SIP-log file If the input parameter is incorrect, the following notification appears: >clearlog: incorrect type of manager <x>
33	removelog >removelog<mngr_dest>	Removes the specified log file. The input parameter specifies the type of logging manager: • 0 (default)—ECR-log file • 1—SIP-log file If the input parameter is incorrect, the following notification appears: >removelog: incorrect type of manager <x>
34	reset2factory >reset2factory	Resets the IP Deskphone to the default setting. See Activating Reset to Factory Setting using SSH_PDT on page 281.

Device configuration file

The following table describes the configuration commands in the device configuration file for alarms, logs and diagnostics.

Table 118: Alarms, logs and diagnostics configuration commands

Component	Flag	Description
PC Port Mirroring parameter which can be modified in the Advanced Diag Tools dialog. See 2.2.1.6	PORT_MIRROR_ENABLE	Determines whether the option can be managed or not. • If PORT_MIRROR_ENABLE is configured as YES, The Port Mirroring prompt in the Advanced Diag Tools dialog is enabled, and you can activate or deactivate the option. • If PORT_MIRROR_ENABLE is configured as NO, the Port Mirroring prompt in the Advanced Diag Tools dialog is

Component	Flag	Description
		disabled (dimmed); the option is deactivated by force and you cannot manage the option. • The values are YES and NO. The default value is NO (disabled).
Memory Monitor. See 2.2.4	MEMCHECK_PERIOD	Determines the time period in seconds when the Memory monitor wakes up (after start-up or the last memory check attempt). The values are 1800 (0.5 hrs) to 86400 (24 hrs). The default value is 86400 (24 hrs).
SIP-traffic monitor	DOS_PACKET_RATE	Determines the maximum number of packets per second that is allowed.
SIP_traffic monitor	DOS_MAX_LIMIT	Specifies how many packets past DOS_PACKET_RATE the IP Deskphone can receive before packets are dropped. If packets are received at a rate of DOS_PACKET_RATE +1, then packets start getting dropped after the time specified in DOS_MAX_LIMIT (in seconds).
SIP-traffic monitor	DOS_LOCK_TIME	Specifies the amount of time (in seconds) the IP Deskphone stops processing packets after DOS_MAX_LIMIT is reached. If DOS_PACKET_RATE is < 1, other values are ignored and packets are not dropped.
Logging System	LOGSIP_ENABLE	Allows the administrator to enable or disable SIP-logging. • If the parameter is YES, the SIP-logging Manager is active and starts logging SIP incoming and outgoing packages into the log files in FFS. • The values are YES and NO. The default value is NO (the

Component	Flag	Description
		manager is not active and the IP Deskphone does not log in SIP incoming and outgoing packages.

Diagnostic Logs

The IP Deskphone supports two types of log files:

- ECR-log
- SIP-log

ECR-log file

The ECR-log file registers and provides detailed information on the errors or bugs that occur during the operation of the IP Deskphone. The ECR-log also contains records indicating some events, such as restart.

Each error is logged as a record. The format of the record is the same regardless of the monitor that generates it or the level of severity of the error. There are three sections to the record.

The first section provides mandatory information for each record including:

- severity level
- severity flag
- time stamp
- software version
- source file information
- error number
- brief description

For example:

```
=== Record #001 === MAJOR SET Logged 01/07/2002 00:34:35 Firmware: 06A5C1Hd10
Description: Task Monitor: the Transport task is suspended
```

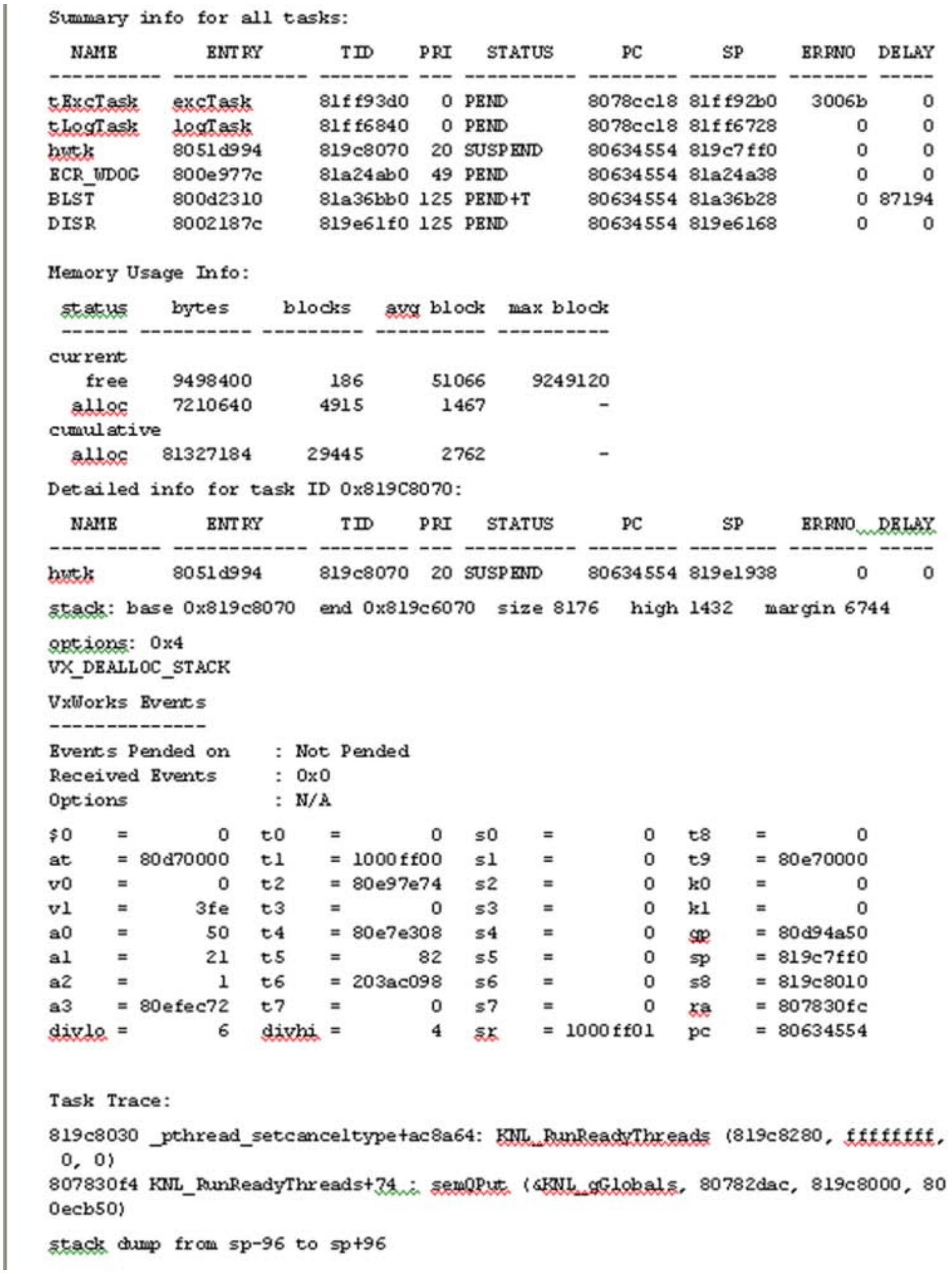
The second section is optional. If the task is registered in the list of stack overflow events, the following may occur:

```
ERROR*ecrStackShow: :StackOverflow: PDT tpStackBase = 0x8194ffa0,
pStackLimit=0x8194bfa0, pStackEnd= 0x8194bfa0 tstack: base 0x8194ffa0 end
0x8194bfa0 size 16368 high 1492 margin 14874
```

The third section includes the supplementary information. The content depends on the flag in the calling function. The flag can be as follows:

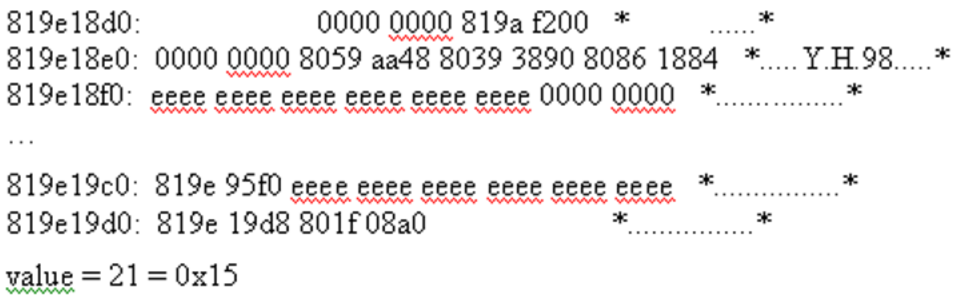
- ECR_LOG_NO_EXTRA_INFO: no supplementary information
- ECR_LOG_TASK_INFO: log task information (ti, tt, the stack information from SP-96 to SP+96)
- ECR_LOG_SUM_TASK_INFO: log summary of each task TCB (i)
- ECR_LOG_MEM_INFO: log memory usage information (memShow)

The following is an example of the supplementary information in the ECR-log file:



```
Summary info for all tasks:

  NAME        ENTRY        TID      PRI   STATUS      PC       SP      ERRNO  DELAY
---------- ------------ -------- --- ---------- -------- -------- ------- -----
tExcTask   excTask      81ff93d0   0 PEND       8078cc18 81ff92b0   3006b     0
tLogTask   logTask      81ff6840   0 PEND       8078cc18 81ff6728       0     0
hwtk       8051d994     819c8070  20 SUSPEND    80634554 819c7ff0       0     0
ECR_WDOG   800e977c     81a24ab0  49 PEND       80634554 81a24a38       0     0
BLST       800d2310     81a36bb0 125 PEND+T     80634554 81a36b28       0 87194
DISR       8002187c     819e61f0 125 PEND       80634554 819e6168       0     0

Memory Usage Info:

 status    bytes     blocks   avg block  max block
 ------ ---------- --------- ---------- ----------
current
   free    9498400       186      51066    9249120
  alloc    7210640      4915       1467          -
cumulative
  alloc   81327184     29445       2762          -

Detailed info for task ID 0x819C8070:

  NAME        ENTRY        TID      PRI   STATUS      PC       SP      ERRNO  DELAY
---------- ------------ -------- --- ---------- -------- -------- ------- -----
hwtk       8051d994     819c8070  20 SUSPEND    80634554 819e1938       0     0

stack: base 0x819c8070  end 0x819c6070  size 8176   high 1432   margin 6744

options: 0x4
VX_DEALLOC_STACK

VxWorks Events
--------------

Events Pended on     : Not Pended
Received Events      : 0x0
Options              : N/A

$0     =        0  t0     =        0  s0     =        0  t8     =        0
at     = 80d70000  t1     = 1000ff00  s1     =        0  t9     = 80e70000
v0     =        0  t2     = 80e97e74  s2     =        0  k0     =        0
v1     =      3fe  t3     =        0  s3     =        0  k1     =        0
a0     =       50  t4     = 80e7e308  s4     =        0  gp     = 80d94a50
a1     =       21  t5     =       82  s5     =        0  sp     = 819c7ff0
a2     =        1  t6     = 203ac098  s6     =        0  s8     = 819c8010
a3     = 80efec72  t7     =        0  s7     =        0  ra     = 807830fc
divlo  =        6  divhi  =        4  sr     = 1000ff01  pc     = 80634554


Task Trace:

819c8030 _pthread_setcanceltype+ac8a64: KNL_RunReadyThreads (819c8280, fffffff,
 0, 0)
807830f4 KNL_RunReadyThreads+74 : semQPut (&KNL_gGlobals, 80782dac, 819c8000, 80
0ecb50)

stack dump from sp-96 to sp+96
```

Figure 85: Example of the supplementary information in the ECR-log file

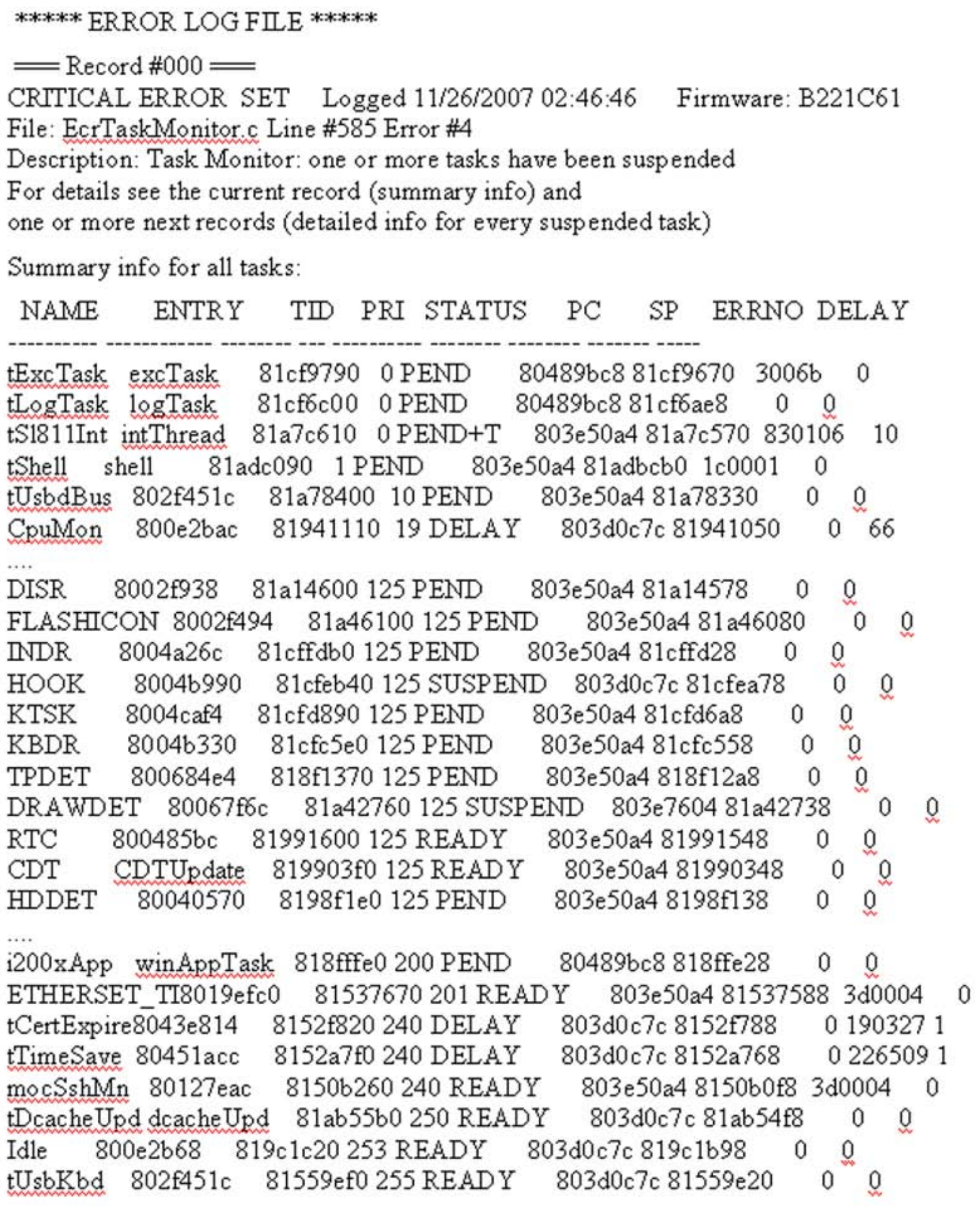


```
819e18d0:                 0000 0000 819a f200   *       ......*
819e18e0:  0000 0000 8059 aa48 8039 3890 8086 1884  *.....Y.H.98.....*
819e18f0:  eeee eeee eeee eeee eeee eeee 0000 0000  *................*
...
819e19c0:  819e 95f0 eeee eeee eeee eeee eeee eeee  *................*
819e19d0:  819e 19d8 801f 08a0                      *................*

value = 21 = 0x15
```

Figure 86: Example of the supplementary information in the ECR-log file (continued)

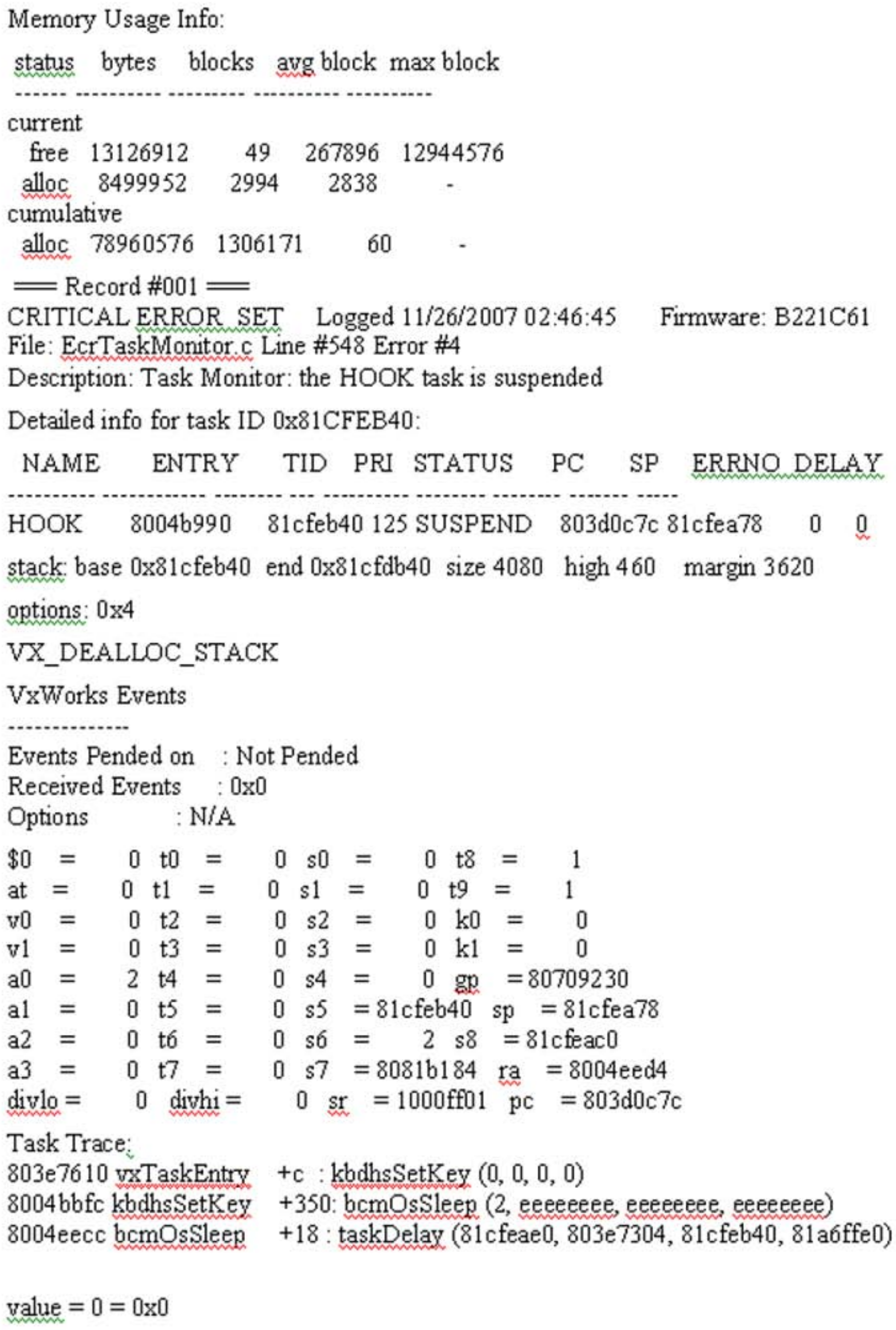
The following is an example of the ECR-log file.

```
PDT>prtlog 0 (-----------------> example of the ECR-log file)
```



```
***** ERROR LOG FILE *****

=== Record #000 ===
CRITICAL ERROR SET    Logged 11/26/2007 02:46:46     Firmware: B221C61
File: EcrTaskMonitor.c Line #585 Error #4
Description: Task Monitor: one or more tasks have been suspended
For details see the current record (summary info) and
one or more next records (detailed info for every suspended task)

Summary info for all tasks:
 NAME       ENTRY      TID    PRI  STATUS     PC       SP    ERRNO  DELAY
---------- ------------ -------- --- ---------- -------- -------- ------- -----
tExcTask   excTask     81cf9790   0 PEND      80489bc8 81cf9670   3006b    0
tLogTask   logTask     81cf6c00   0 PEND      80489bc8 81cf6ae8       0    0
tS1811Int  intThread   81a7c610   0 PEND+T    803e50a4 81a7c570  830106   10
tShell     shell       81adc090   1 PEND      803e50a4 81adbcb0  1c0001    0
tUsbdBus   802f451c    81a78400  10 PEND      803e50a4 81a78330       0    0
CpuMon     800e2bac    81941110  19 DELAY     803d0c7c 81941050       0   66
....
DISR       8002f938    81a14600 125 PEND      803e50a4 81a14578       0    0
FLASHICON  8002f494    81a46100 125 PEND      803e50a4 81a46080       0    0
INDR       8004a26c    81cffdb0 125 PEND      803e50a4 81cffd28       0    0
HOOK       8004b990    81cfeb40 125 SUSPEND   803d0c7c 81cfea78       0    0
KTSK       8004caf4    81cfd890 125 PEND      803e50a4 81cfd6a8       0    0
KBDR       8004b330    81cfc5e0 125 PEND      803e50a4 81cfc558       0    0
TPDET      800684e4    818f1370 125 PEND      803e50a4 818f12a8       0    0
DRAWDET    80067f6c    81a42760 125 SUSPEND   803e7604 81a42738       0    0
RTC        800485bc    81991600 125 READY     803e50a4 81991548       0    0
CDT        CDTUpdate   819903f0 125 READY     803e50a4 81990348       0    0
HDDET      80040570    8198f1e0 125 PEND      803e50a4 8198f138       0    0
....
i200xApp   winAppTask  818fffe0 200 PEND      80489bc8 818ffe28       0    0
ETHERSET_TI8019efc0    81537670 201 READY     803e50a4 81537588  3d0004    0
tCertExpire8043e814    8152f820 240 DELAY     803d0c7c 8152f788       0 190327 1
tTimeSave  80451acc    8152a7f0 240 DELAY     803d0c7c 8152a768       0 226509 1
mocSshMn   80127eac    8150b260 240 READY     803e50a4 8150b0f8  3d0004    0
tDcacheUpd dcacheUpd   81ab55b0 250 READY     803d0c7c 81ab54f8       0    0
Idle       800e2b68    819c1c20 253 READY     803d0c7c 819c1b98       0    0
tUsbKbd    802f451c    81559ef0 255 READY     803d0c7c 81559e20       0    0
```

Figure 87: Example of the ECR-log file



```
Memory Usage Info:
 status   bytes    blocks   avg block  max block
 ------ ---------- --------- ---------- ----------
current
   free  13126912      49    267896   12944576
  alloc   8499952     2994     2838        -
cumulative
  alloc  78960576   1306171       60        -
 === Record #001 ===
CRITICAL ERROR_SET    Logged 11/26/2007 02:46:45     Firmware: B221C61
File: EcrTaskMonitor.c Line #548 Error #4
Description: Task Monitor: the HOOK task is suspended
Detailed info for task ID 0x81CFEB40:
  NAME      ENTRY      TID   PRI  STATUS     PC      SP     ERRNO  DELAY
---------- ----------- -------- --- --------- -------- -------- ------- -----
HOOK      8004b990    81cfeb40 125 SUSPEND   803d0c7c 81cfea78      0     0
stack: base 0x81cfeb40  end 0x81cfdb40  size 4080  high 460   margin 3620
options: 0x4
VX_DEALLOC_STACK
VxWorks Events
--------------
Events Pended on     : Not Pended
Received Events      : 0x0
Options              : N/A
$0   =      0  t0   =      0  s0   =      0  t8   =      1
at   =     0  t1   =     0  s1   =     0  t9   =     1
v0   =      0  t2   =      0  s2   =      0  k0   =      0
v1   =      0  t3   =      0  s3   =      0  k1   =      0
a0   =      2  t4   =      0  s4   =      0  gp   = 80709230
a1   =      0  t5   =      0  s5   = 81cfeb40  sp   = 81cfea78
a2   =      0  t6   =      0  s6   =      2  s8   = 81cfeac0
a3   =      0  t7   =      0  s7   = 8081b184  ra   = 8004eed4
divlo =      0  divhi =      0  sr   = 1000ff01  pc   = 803d0c7c
Task Trace:
803e7610 vxTaskEntry    +c   : kbdhsSetKey (0, 0, 0, 0)
8004bbfc kbdhsSetKey    +350: bcmOsSleep (2, eeeeeeee, eeeeeeee, eeeeeeee)
8004eecc bcmOsSleep     +18 : taskDelay (81cfeae0, 803e7304, 81cfeb40, 81a6ffe0)

value = 0 = 0x0
```

Figure 88: Example of the ECR-log file (continued)

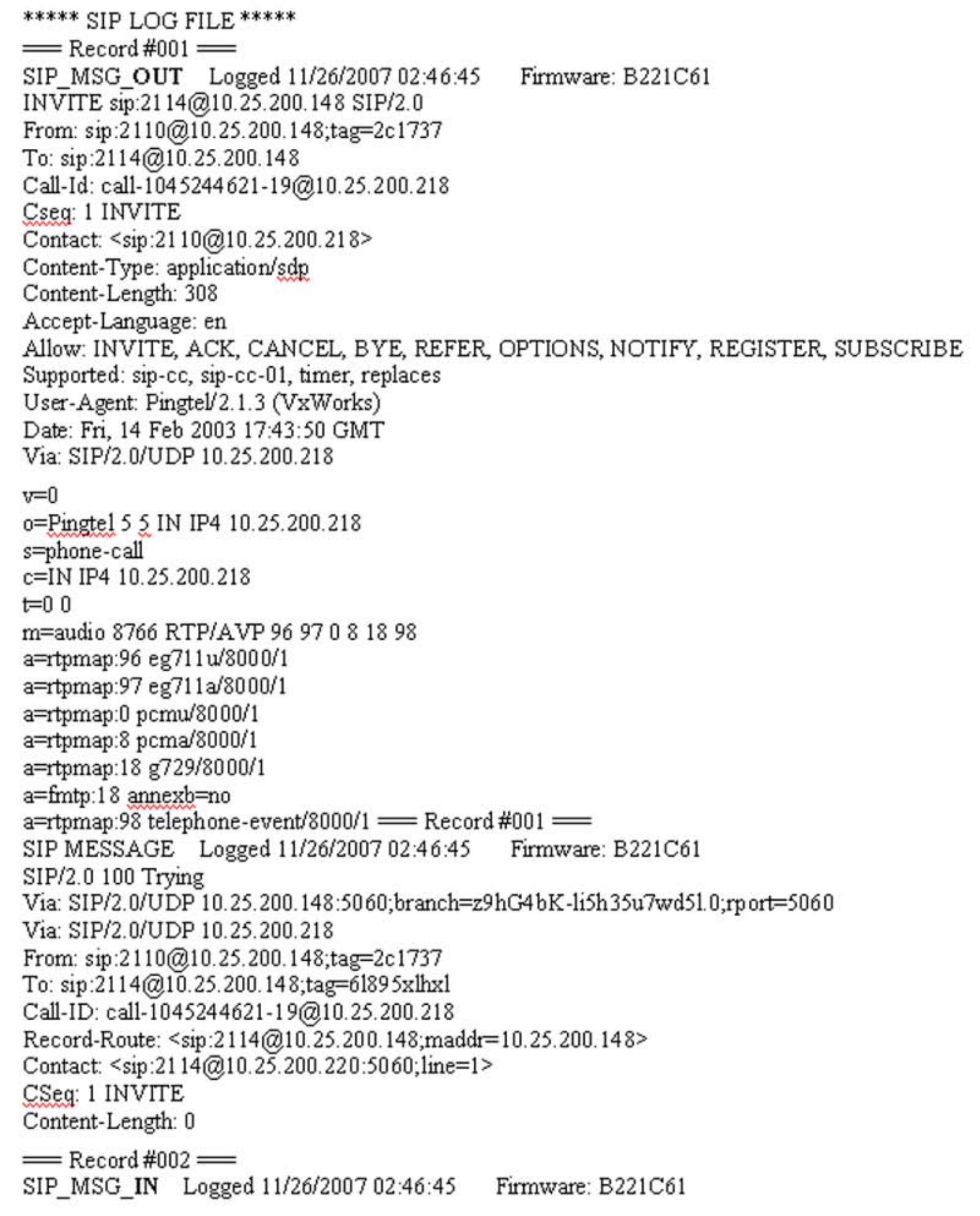
SIP-log file

The SIP-log file registers incoming and outgoing SIP-packages, and each package is logged as a record. There are two sections:

- The first section requires mandatory information for each record including:
    - type of the package (incoming or outgoing)
    - time stamp
    - software version
- The second section contains the content of the package in the text format.

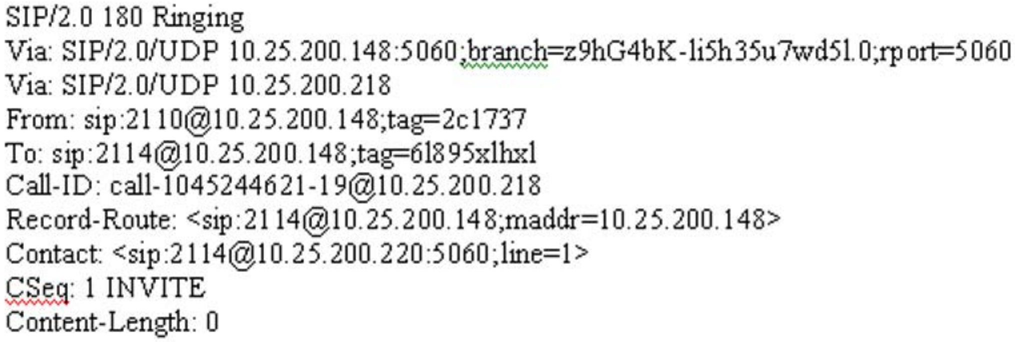
The following is an example of the SIP-log file.

```
PDT>prtlog 1 (----------------->  example of the SIP-log file)
```



```
***** SIP LOG FILE *****
=== Record #001 ===
SIP_MSG_OUT    Logged 11/26/2007 02:46:45       Firmware: B221C61
INVITE sip:2114@10.25.200.148 SIP/2.0
From: sip:2110@10.25.200.148;tag=2c1737
To: sip:2114@10.25.200.148
Call-Id: call-1045244621-19@10.25.200.218
Cseq: 1 INVITE
Contact: <sip:2110@10.25.200.218>
Content-Type: application/sdp
Content-Length: 308
Accept-Language: en
Allow: INVITE, ACK, CANCEL, BYE, REFER, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE
Supported: sip-cc, sip-cc-01, timer, replaces
User-Agent: Pingtel/2.1.3 (VxWorks)
Date: Fri, 14 Feb 2003 17:43:50 GMT
Via: SIP/2.0/UDP 10.25.200.218

v=0
o=Pingtel 5 5 IN IP4 10.25.200.218
s=phone-call
c=IN IP4 10.25.200.218
t=0 0
m=audio 8766 RTP/AVP 96 97 0 8 18 98
a=rtpmap:96 eg711u/8000/1
a=rtpmap:97 eg711a/8000/1
a=rtpmap:0 pcmu/8000/1
a=rtpmap:8 pcma/8000/1
a=rtpmap:18 g729/8000/1
a=fmtp:18 annexb=no
a=rtpmap:98 telephone-event/8000/1 === Record #001 ===
SIP MESSAGE    Logged 11/26/2007 02:46:45       Firmware: B221C61
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.25.200.148:5060;branch=z9hG4bK-li5h35u7wd5l.0;rport=5060
Via: SIP/2.0/UDP 10.25.200.218
From: sip:2110@10.25.200.148;tag=2c1737
To: sip:2114@10.25.200.148;tag=61895xlhxl
Call-ID: call-1045244621-19@10.25.200.218
Record-Route: <sip:2114@10.25.200.148;maddr=10.25.200.148>
Contact: <sip:2114@10.25.200.220:5060;line=1>
CSeq: 1 INVITE
Content-Length: 0

=== Record #002 ===
SIP_MSG_IN    Logged 11/26/2007 02:46:45       Firmware: B221C61
```

Figure 89: Example of the SIP-log file



```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 10.25.200.148:5060;branch=z9hG4bK-li5h35u7wd51.0;rport=5060
Via: SIP/2.0/UDP 10.25.200.218
From: sip:2110@10.25.200.148;tag=2c1737
To: sip:2114@10.25.200.148;tag=61895xlhxl
Call-ID: call-1045244621-19@10.25.200.218
Record-Route: <sip:2114@10.25.200.148;maddr=10.25.200.148>
Contact: <sip:2114@10.25.200.220:5060;line=1>
CSeq: 1 INVITE
Content-Length: 0
```

Figure 90: Example of the SIP-log file (continued)

SIP Software Web Manager

The SIP Software Web Manager stores all the diagnostic logs at:

`/usr/local/sipx/var/log/sipxpbx`

HTTP server logs

To view all logs related to the http server, go to:

```
/usr/local/sipx/var/log/sipxbx/httpd_access_log /usr/local/sipx/var/
log/sipxbx/httpd_error_log /usr/local/sipx/var/log/sipxbx/
httpd_rewrite_log
```

Installation logs

To view the log of all the installed packages, go to:

`/root/install.log`

To view all system messages during startup, go to:

`/var/log/messages`

Configuration server logs

To view all logs related to the configuration server, go to:

```
/usr/local/sipx/var/log/sipxbx/sipxconfig.log /usr/local/sipx/var/
log/sipxbx/sipxconfig.logins.log
```

Backup and restore logs

For more information on backup and restore logs, see Using the SIP Software Web Manager (NN43112-500).

Login logs

To view the login history in the SIP Software Web Manager, go to:

Diagnostics->Login History

For more information on the SIP Software Web Manager, see Using the SIP Software Web Manager (NN43112-500).

PC Client Softphone interworking with the IP Deskphone

If the user does not have access to the pre-authorization configurations in the Feature Options menu, the feature is not enabled. You must verify the device configurations and enable the interworking feature so that the user can access the pre-grant authorization configuration and the IP Deskphone can auto-answer calls from authorized users or user groups. For more information, see Configuration of the PC Client Softphone on page 277.

If the call is being received, but is not being automatically answered in a Click-to-Answer scenario, the user must verify that the user making the request is an authorized user. For more information, see Pre-granting authorization for the Answer-Mode on page 271.

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